Sonnox

Fraunhofer IIS

Pro-Codec

User Guide

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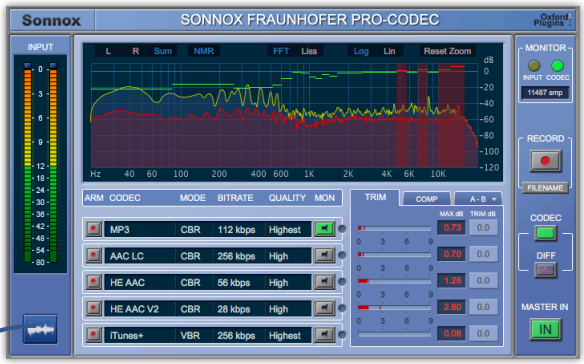
1 Introduction

In the past it has not been possible to critically audition and then encode multiple formats in real time within a host DAW environment. The revolutionary Sonnox Fraunhofer Pro-Codec Plug-In now makes this possible.

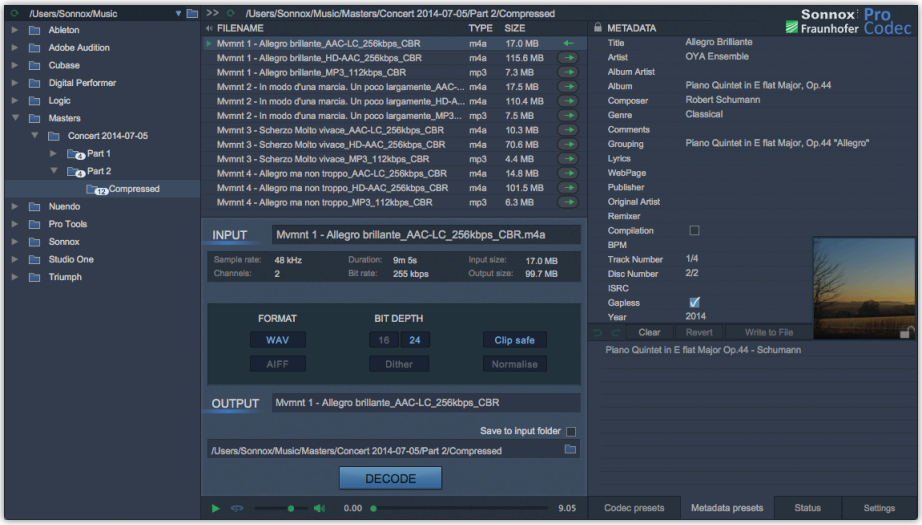
The Pro-Codec Plug-In is designed for the real-time auditioning of audio signals by the Fraunhofer codecs. The Plug-In also provides the ability to record the real-time encoded data to disk (in the form of MP3 and M4A files).

This enables mixing engineers to produce compensated mixes optimised towards specific target codecs, thereby ensuring maximum fidelity. Similarly, mastering engineers may audition material in the final format, and produce compensated, optimised masters for final encoding and distribution.

The included Pro-Codec Manager is a powerful desktop application designed for offline batch encoding, decoding and metadata editing of your encoded files.



Click to open the Manager (or open it directly from disk)



2 Terminology

Bit rate

This is the number of bits transmitted or processed per unit of time, expressed here in kilobits per second (kbps).

CBR

Constant Bit Rate –- bit rate remains the same for the duration of the audio signal.

VBR

Variable Bit Rate –- bit rate changes over time, based on the complexity of the signal. In this case, the VBR value expressed is an average over the length of the file.

Encoder

In this context, an encoder converts an uncompressed LPCM audio stream into a smaller, compressed format through the use of lossy compression. For HD-AAC, the encoding process is lossless to the target bit depth.

Decoder

In this context, a decoder converts a compressed audio stream back into an uncompressed LPCM audio stream.

Metadata

This is categorical information about a file, such as the song Title, Artist and Album.

Tag A tag is one specific piece of metadata, such as the song Title.

3 Supported Codecs and Formats

Major Codecs Supported

- MP3
- AAC-LC
- HE-AAC
- HE-AAC v2
- MPEG Surround
- Apple AAC (iTunes Plus; Mastered for iTunes) (Mac only)

Lossless Codecs Supported

- HD-AAC (16/24 bits up to 192 kHz)

Formats Supported

Mono

MP3, AAC-LC, HE-AAC and HD-AAC

Stereo

MP3, AAC-LC, HE-AAC, HE-AACv2 and HD-AAC, Apple AAC (iTunes Plus; Mastered for iTunes) (Mac only)

5.1 AAC-LC, HE-AAC and HD-AAC, MPEG Surround

4 Summary of Codec Features and Applications

MP3

General purpose audio codec, compatible with most players.

Typical bit rate: 128 kbps (stereo)

AAC-LC (AAC Low Complexity)

High performance audio codec for excellent audio quality at low bit rates. Used for ISDB television (Japan).

Typical bit rate: 128 kbps (stereo)

HE-AAC (High Efficiency AAC)

High performance audio codec for good quality at bit rates of 32–48 kbps per channel. Used for XM Radio, mobile music downloads and Digital Radio Mondiale.

Typical bit rate: 64 kbps (stereo)

HE-AAC v2

High performance audio codec for good quality at bit rates of 16–24 kbps per channel. Used for 3GPP music download, Digital Radio DAB+, and internet radio streaming to mobile devices (e.g. iPhone).

Typical bit rate: 48 kbps (stereo)

HD-AAC (High Definition AAC)

Lossless audio codec with an optional lossy core. Used for music distribution and archival.

Typical bit rate: roughly half the bit rate of a 16-bit uncompressed WAV file.

Apple AAC

High performance audio codec for excellent audio quality at low bit rates. Used for Apple iPod, iTunes etc.

Typical bit rate: 256 kbps VBR (stereo)

4.1 Apple AAC – iTunes Plus (Mastered for iTunes) (Mac only)

The addition of this Apple AAC codec allows real-time audition of files produced under the Mastered for iTunes initiative that are destined for the iTunes store.

This codec is available for Mac computers only. It uses the same Apple codecs and resamplers that are used for the current iTunes Catalogue. It very specifically resamples to 44.1 kHz if necessary, and encodes as AAC-LC, 256 kbps, variable bit rate and highest quality, which are the settings used for the iTunes catalogue. For online audition, the signal is then resampled back to the host DAW sample rate if necessary.

This option is available only for stereo configuration. There are no options to change the Apple codec settings because this would not be representative of the iTunes Plus standard. The Apple codecs and re-samplers are components of the Mac OSX operating system, so can be upgraded during an OSX update.

Note:
On Windows computers, the closest approximation to the iTunes Plus codec is the Fraunhofer AAC-LC codec set to highest quality VBR at 256 kbps.

4.2 MPEG Surround (Fraunhofer)

MPEG Surround is a feature-rich open standard compression technique for multi-channel audio signals. Operating on top of any core audio codec — including AAC, HE-AAC and MPEG Layer 2 – the system provides a comprehensive feature set, including highest surround and stereo audio quality, multi-channel support at stereo bit rates and virtualised binaural multi-channel sound reproduction capability for stereo headphones.

4.3 HD-AAC (Fraunhofer)

The HD-AAC codec from Fraunhofer has a very clever feature; the single compressed lossless file includes a lossy core channel. It therefore acts as a lossless archival format, a lossless distribution format for the masters, and a final playback format for both lossless and lossy decoders – and all of this in a single file.

In the case of HD-AAC playback, if the decoder in your player has full HD-AAC capability, you will be able to listen to a perfect replication of the original WAV file. However, the same HD-AAC file will still play through a decoder that does not have HD-AAC capability, and instead you will be listening to the embedded lossy AAC channel.

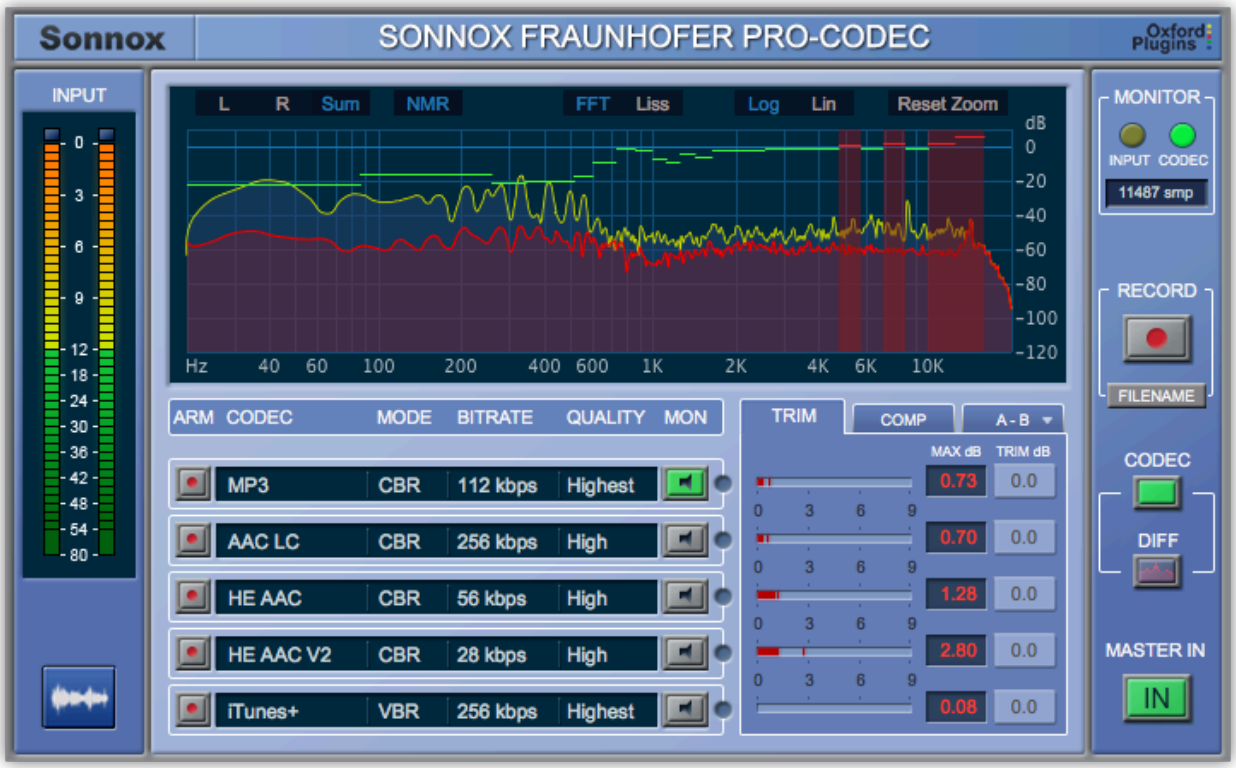
The HD-AAC codec now supports 16-bit files in addition to 24-bit data.

4.4 AAC-LC (Fraunhofer)

The AAC-LC codec now supports a 96 kHz session sample rate and a VBR mode at 256 kbps.

5 The Pro-Codec Plug-In

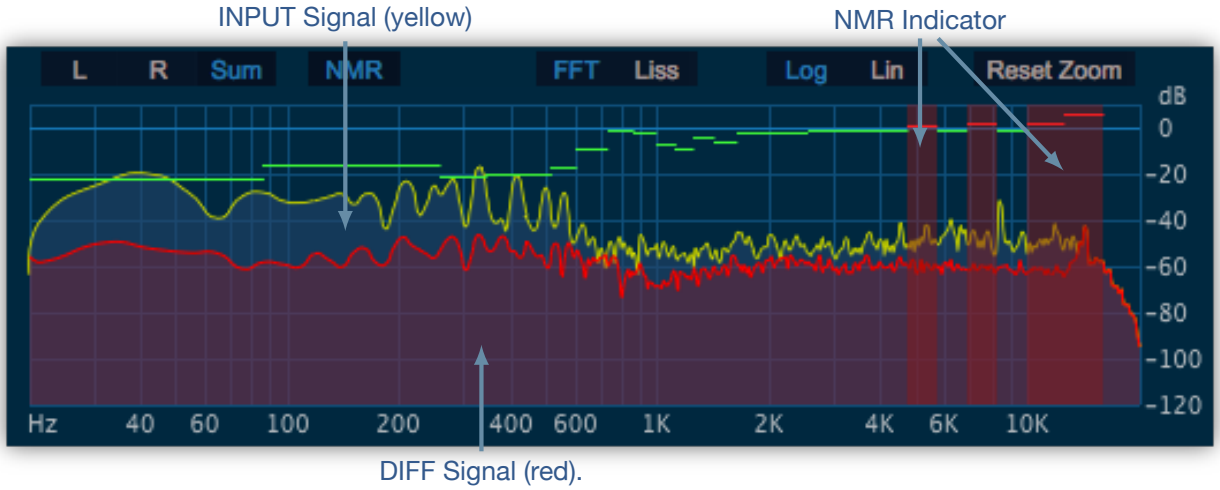
5.1 User Interface and Workflow Overview



The primary purpose of the Pro-Codec Plug-In is to provide a way of monitoring your host session material using the highly regarded Fraunhofer encoders, without the need for offline processing. By auditioning and comparing codecs in real-time, the Pro-Codec Plug-In allows you to make executive decisions about the best codecs to use. When this decision is made, bounce your mix/master to WAV/AIFF and encode this file using the included Manager application.

The latest version of this plug-in also includes the Apple AAC codec (Mac only), which allows users to directly audition audio that has been produced according to the ‘Mastered for iTunes’ specification (also known as ‘iTunes Plus’).

FFT (Fast Fourier Transform) display



DIFF Signal

Red trace. This is the difference between the input and output signals.

NMR

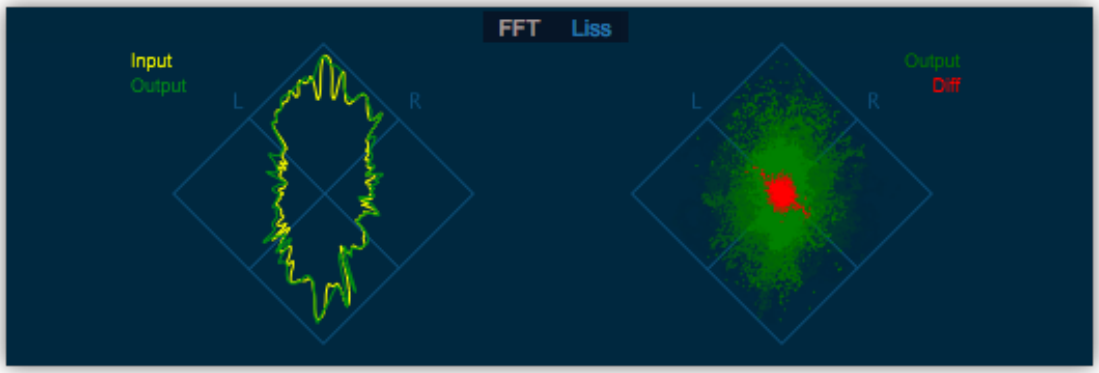
Green/Red Trace. Noise to Mask Ratio is an indication of the audibility of codec-induced artefacts. See section 5.3.1 for more information.

To **ZOOM** the display, Alt + drag (Mac and Windows).

For a stereo signal, you can use the buttons at the top-left of the display to display only the LEFT or RIGHT, or the SUM of both left and right, or hide/show the NMR indicator line.

LISS (Lissajous) display

Click the **LISS** button at the top of the display area to switch to a lissajous display. The left lissajous display shows the input and output envelopes, while the right display shows the stereo spread of the codec and diff signals. **ALT + drag** in the right display to zoom.

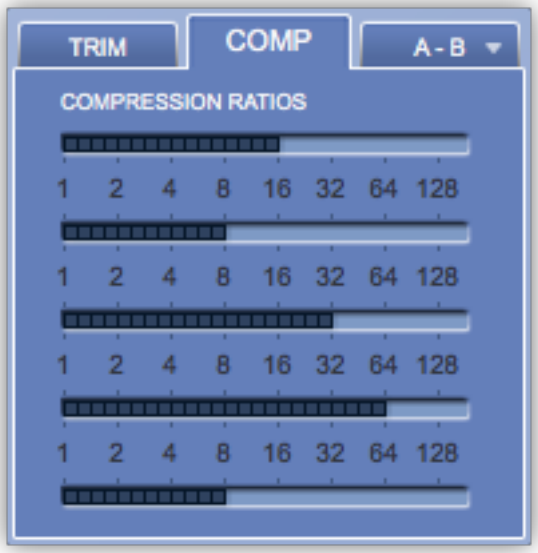


Beneath the FFT/LISS display, the CODEC LIST enables the selection of up to five different codecs. Use the **MON** buttons to switch the real-time monitor path between the loaded codes. Use the **ARM** buttons to arm a codec for online (real-time) recording to an MP3 or M4A file.

ARM CODEC MODE BITRATE QUALITY MON
MP3 CBR 112 kbps Highest
AAC LC CBR 256 kbps High
HE AAC CBR 56 kbps High
HE AAC V2 CBR 28 kbps High
iTunes+ VBR 256 kbps Highest

To the right of the codec list are three tabbed windows:

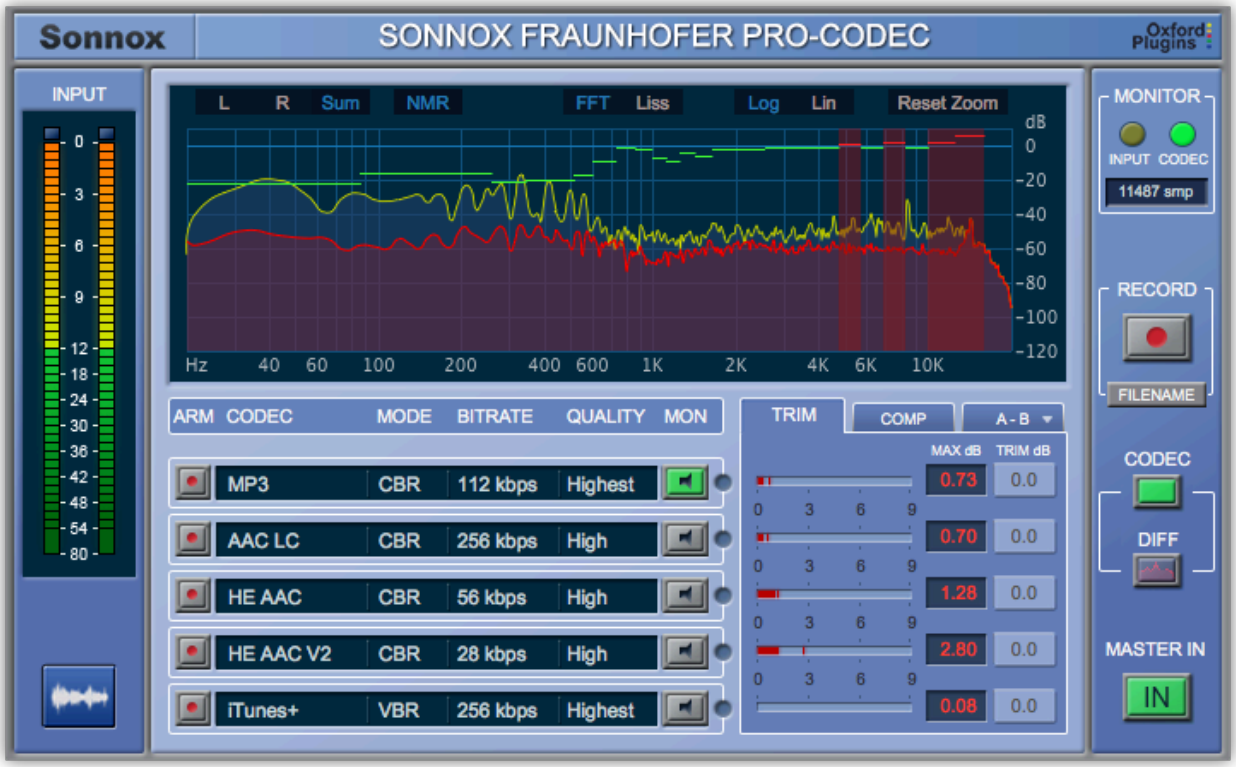
BITSTREAM Level / TRIM – These meters show the decoded sample level for each codec.



COMP – These meters show the data compression ratio for each codec.

A-B (A-B-X) – This tab provides functions for A-B comparison and A-B-X listening tests. Click the down-arrow to switch the tab between A-B and A-B-X mode.

5.2 Input and Output Panels



The left hand panel of the plug-in features a large input signal meter and a button to launch the included Manager application for offline encoding and metadata editing. See section 7 for details. These meters show 2 dB per LED below -12 dBFs, and 0.5dB per LED above -12 dBFs.

The output panel on the right of the GUI includes a **MASTER IN** button, for glitch-free switching between the monitor output of the plug-in (usually a codec output) and the input signal. The signal you are monitoring is indicated by two **LEDs** at the top of the panel, designated INPUT (yellow) and CODEC (green).

A field below the **MONITOR LEDs** displays the **TOTAL DELAY** through the plug-in, which is determined by the maximum of each individual codec delay. Click the value field to toggle between milliseconds and samples. The delay value is immediately updated in the GUI when a new codec or other setting is selected. Depending on the host behaviour, a new value may not be available for delay compensation until the plug-in is removed and re-instantiated.

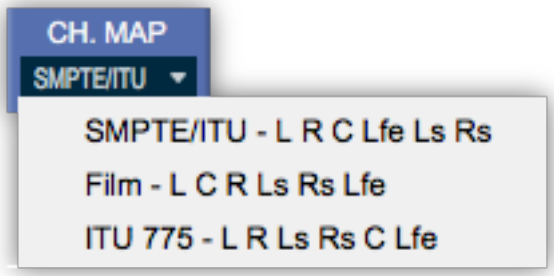
The listen section of the output panel defaults to the **CODEC** monitor output, but also contains an option to listen to the difference between the input and the processed output by selecting the **DIFF** button. This is particularly useful for monitoring the audio that is

removed by the encoding process.

Enable the **RECORD** button to begin real-time encoding to disk for all **ARM**ed codecs. Click the **FILENAME** button to specify where encoded files are written to when recording, and how they are named.

5.2.1 Surround Channel Mapping

When the Pro-Codec Plug-In is inserted on a 6-channel track, the surround channel order menu button appears at the bottom of the INPUT panel.



Use these options to match the input/output channel mapping of the plug-in to the channel mapping used in your host DAW. If you use SMPTE/ITU or Film channel order, simply select the appropriate option.

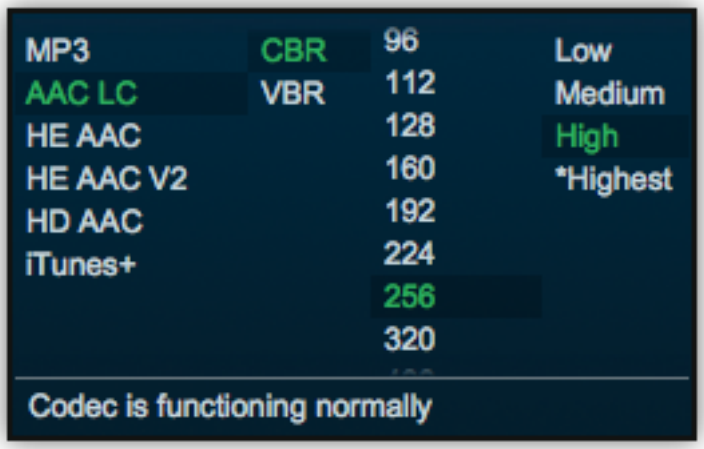
5.3 Online Auditioning and Configuring Codecs

The Pro-Codec Plug-In enables Fraunhofer codecs to be auditioned, compared and the encoded audio written to disk, all while the host workstation is operating.

To add a codec, or change it settings, just click on the codec row in the codec list:

ARM CODEC MODE BITRATE QUALITY MON
MP3 CBR 112 kbps Highest
AAC LC CBR 256 kbps High
HE AAC CBR 56 kbps High
HE AAC V2 CBR 28 kbps High
iTunes+ VBR 256 kbps Highest

This will open the codec settings selector:



When a codec, bit rate or mode setting is changed, the real-time output of the plug-in immediately reflects this change. To close the selector, simply move your mouse cursor elsewhere or press the **ESCAPE** key.

Some items may be marked with an asterisk to indicate that the setting will require some form of resampling. Details will be shown in the lines at the bottom of the selector panel.

Some combinations of bit rate, quality setting and sample rate might cause an encoder to down-sample internally (especially at the highest quality settings). In this case, if the stream is decoded for auditioning, the number of samples per second produced will not match the session sample rate and there will be a pitch shift.

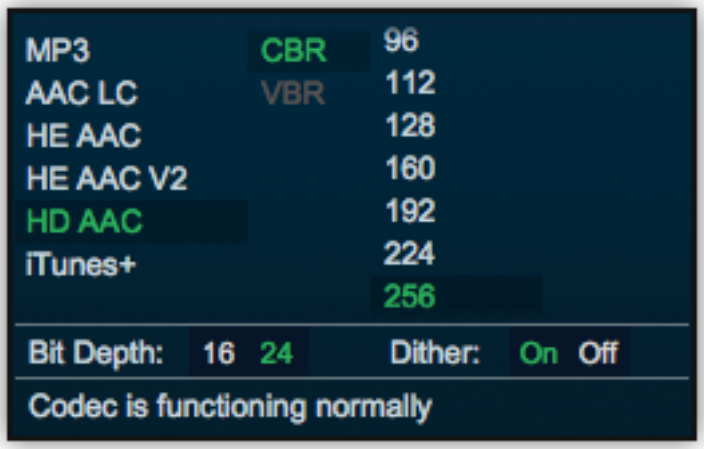
To work around this problem, when one of these special combinations is selected, the plug-in creates two versions of the codec; one operating at lower quality for auditioning (so that internal down-sampling does not take place), and one operating at the selected quality, which will be used only for writing to file if the user drops into online record mode.

In a 44.1 kHz session, selecting Highest quality for these codec settings will result in the encoded output file having an internal sample rate of 48 kHz.

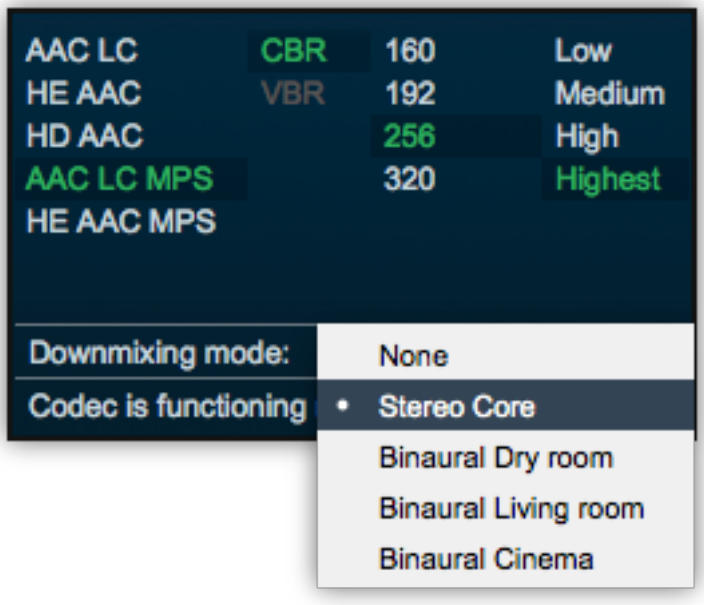
At higher session sample rates, the codec settings selector shows when down-sampling is occurring. In this case, the input audio is down-sampled prior to encoding, to bring the sample rate into the range supported by the codec. The decoded audition signal is correspondingly up-sampled.

Note:
The encoded file's sample rate is derived from the session sample rate, the codec supported sample rate, the selected codec bit-rate, the selected codec quality setting and the channel configuration. Hence, it is not a simple process to choose a desired encoded file sample rate. For more details, see section **9.3.3**.

The codec settings selector also provides additional options for HD-AAC, and for auditioning down-mixing modes for the 5.1 MPS codecs:



HD-AAC



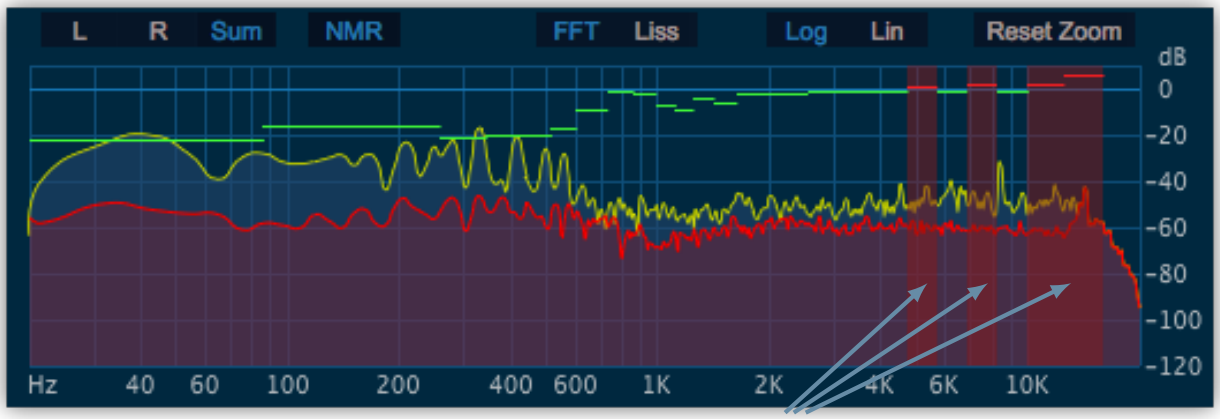
AAC-LC MPS (5.1) down-mixing modes

5.3.1 The NMR Indicator

Also featured in the FFT display is the **NMR** (Noise to Mask Ratio) indicator line. This provides an indication of the frequency areas where the difference between the codec output and the original input might be audible. The **NMR** indicator can be disabled by clicking on the **NMR** button (above the display).

All lossy codecs will produce a very slightly different output from their input. Sometimes, if you listen to the difference signal it can sound significant; however, the very nature of a perceptual coder is that this difference signal should be inaudible (i.e. masked by the output signal). You can choose to trade off more data compression (and smaller files) against increased audibility of artefacts and codec-induced noise. Theory states that this codec-induced noise should be inaudible when the NMR indicator is green.

Under some circumstances (codec, frequency and input signal dependent) this **NMR** can turn red, indicating that the induced artefacts are potentially audible.



There are potentially audible artefacts in these frequency bands.

The listening environment, and the training and sensitivity of the listener's ears are obvious and variable factors to take into account regarding the audibility of these artefacts.

The **NMR** display is a useful indication when auditioning the MP3 or AAC codec families. If there are areas where the **NMR** line is above 0 dB, those are the frequency areas of interest. The display will highlight these frequency areas in red. It is possible, for example, to make very quick comparisons of a selection of different bit rates while auditioning music.

The **NMR** calculation is less accurate for parametric codecs (these use enhancements such as Parametric Stereo or Spectral Band Replication). HE-AAC and HE-AACv2 use parametric enhancements to achieve very high compression ratios. The **NMR** indicator is enabled for these codecs by default, because it can still give an indication to the engineer

of frequency areas that might require examination.

The **NMR** indicator is not present when HD-AAC is selected. Also, the NMR display is less accurate when SUM is selected; it is most accurate with single audio streams.

> **Note:**
> The difference signal used in the **NMR** calculation is absolute. For some low level signals the **NMR** might be relatively high, but any difference will be inaudible due to the low signal level.

The **NMR** indicator is only valid for session sample rates 44.1, 48, 88.2, 96, 176.4 and 192kHz. In particular, it is not supported for a 32 kHz sample rate. The NMR indicator is not valid and is not displayed for frequencies above 16kHz.

5.3.2 Downsampling When Using Higher Sample Rates

There are several circumstances when the plug-in internally down-samples. One example is where the session sample rate is higher than the highest sample rate that the selected codec supports. In this case, the audio is down-sampled for the codec, and then up-sampled for audition. In this case, a down-sampled signal is available in the codec for writing to a file in online encode mode.

This is not the case in the Manager application. The user cannot offline encode a WAV file whose channel configuration and sample-rate is not a match for the codec settings. This is because of possible indeterminate behaviour if the configurations are not compatible. This is particularly important if you are working with higher sample rate files.

If you are working at higher sample rates, the options are exemplified by:

1. Master at high sample rate to produce a gold master, then audition and online encode at the same high sample rate.
2. Master at high sample rate and produce a gold master. Down-sample to 44.1 kHz (for CD). Load CD master session and audition codecs, then encode the CD master with the Manager application.

5.3.3 Overloading Pre-Codec (Codec Headroom)

It is possible, but not desirable, that the plug-in will be fed by an overloading input (sample values greater than 0 dBFS). In this case, the input meters will show clipping. This is separate to the possibility that a lossy encoder given a non-overloading input could produce a bitstream that overloads a decoder output.

If an overloading input is presented to lossy codecs such as AAC-LC, the codec output will endeavour to follow the input as accurately as possible, using internal headroom in both the encoder and decoder. Bitstream Level **Max dB** will accurately indicate the maximum level of the signal, and the signal will not clip in the plug-in.

Note:
This is not true for the HD-AAC codec (lossless), which does not have internal headroom for the lossless layer.

5.3.4 Overloading Post-Codec (Audition Level Matching)

As discussed above, any form of filtering (including encoding and decoding) can increase the peak level of a signal. Some codecs such as HE-AAC can add several dBs, hence the provision of Bitstream Level **Trim** to avoid post-decoder clipping in the audition chain. Up to five codecs can be selected for audition, and each could require a different trim setting. To keep the audition process palatable, the trim is reversed on each channel before the audition point, so that switching between codecs does not introduce audition level changes.

Note:
However, this means that you are not auditioning the true level at the input of the encoder. If there are significant trim values involved, it is possible that the encoded files will demonstrate differing average levels when played back, and this may be undesirable.

The Pro-Codec plug-in provides the ability to observe and monitor bitstream level behaviour, and tools for the user to mitigate this behaviour. It is convenient to monitor many codecs simultaneously for comparison, but it might be best practice to treat each codec individually when producing a final master. In this way any level change can be kept in the master mix, rather than in the plug-in.

5.4 Bitstream Level (TRIM Tab)

Any form of filtering can, under certain circumstances, increase the peak level of the signal. Therefore, if your input is hot, the encoding process can produce overloads that are not seen until the signal is decoded. This effect can be very easily demonstrated with the Pro-Codec Plug-In.

The Bitstream Level **TRIM** tab can be used to monitor this potential for overload. The post-decoder level of each selected codec is continually monitored, and a small meter indicates levels greater than 0 dBFS. There is a peak-hold text display (**MAX dB**) to show the maximum post-decoder level achieved. This text is white if below 0 dBFS and turns red on overload. Clicking on this display resets the overload condition.

The **TRIM dB** control allows you to trim the input level to the codec to prevent these overloads. The gain reduction is applied pre-encoding, and if non-zero the trim value is displayed darker. In order to allow equal loudness listening comparison and difference signal generation, the trim is applied before encoding and inversely applied after the decoder output. This way the encoded bitstream is reduced in level, but the auditioned signal remains at the same level as the input signal.

The **TRIM dB** controls can be automated for use with online encoding. For convenient trimming, the value in the **MAX dB** window can be dragged and dropped onto the **TRIM dB** cell. The **MAX dB** value can be repeatedly dropped onto the **TRIM** cell; each time the value is subtracted from the current **TRIM** value. In the screenshot above, codec 1 is overloading by 0.73 dB. Selecting and dragging the value from **MAX dB** to **TRIM dB** will set the latter to -0.73 dB.

This post-decoder level overload monitoring is present even if the **TRIM** tab is deselected via permanently visible overload LEDs to the right of each of the five codec selection slots. Clicking on an overload LED resets the overload condition for that codec.

It is important to note that the indicated Bitstream metering and overload LEDs are provided as a guide only. If a segment of audio is being cycled, these maximum values are not exactly repeatable from one cycle to the next. Furthermore, in some host applications, repeatedly rewinding to the beginning of the track will also result in

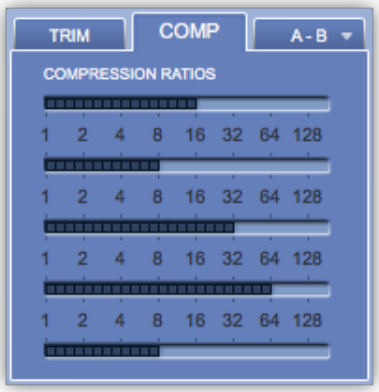
unrepeatable maximum values. The discrepancies in both cases should be minor, and not all source audio and not all codecs will be affected. So, it is not always possible to trim exactly while auditioning, because a subsequent pass might have a slightly different maximum value.

There is a technical reason for this, stemming from the difficulty of synchronizing the start point of the buffer that is used to input data to the codecs.

Note:
In some situations this will be important and, for those occasions, we recommend encoding the file offline using the Manager application. Offline encoding is exactly repeatable however many times the file is encoded.

5.5 Data Compression Factor (COMP Tab)

The **COMP** tab shows the data compression ratio for each of the selected codecs.



All of the available codecs compress an audio data bitstream. Different codecs compress using different algorithms, and will compress to very different degrees.

HE-AAC is optimized for the best quality at very low bit rates, and will produce very small files (potentially around one hundredth of the input file size). HD-AAC, in contrast, is not so effective at compressing data, and will produce files that are around half the size of the input. The familiar MP3 codecs produce compression factors of typically around 12.

A yellow asterisk (if present) indicates that the figure is approximate. This will be the case for HD-AAC and variable bit rate (VBR) encoders.

Note:
Remember that using perceptual coding is always a trade-off between more accurate reproduction of the original on the one hand, against a higher compression ratio and smaller files on the other hand.

5.6 A-B Auditioning and A-B-X Testing (A-B Tab)

5.6.1 A-B Auditioning

For simple and quick auditioning use AB Mode. Control is via the A-B tab.

Clicking in the A or B columns will assign a codec to the A or B audition channels. Clicking in the B column adjacent to a selected A codec will automatically set the B column to the input signal (I/P). To revert to a codec for the B column, click on a cell non-adjacent to the A codec.

The **A-B** buttons toggle between the codec selected in the A column and the codec (or input) selected in the B column. This is ideal for critical listening tests.

The **HIDE** button makes the current selection anonymous. The **A-B** buttons still toggles between the two column selections.

The **BOTH** button displays both of the two **DIFF** signals (the currently auditioning signal is coloured green).

5.6.2 A-B-X Testing

For more detailed and statistical listening tests, use **A-B-X** mode (for the principles behind A-B-X testing, there is a good Wikipedia entry). Select the **A-B-X** tab via the down-arrow on the tab.

Unlike hardware A-B-X systems, there is no switch delay or switching glitches in this implementation.

The two **A-B** assignment columns are used in a similar fashion to **A-B** auditioning, but the column headers are now buttons. There is also an additional button labelled **X**.

Although the three **A-B-X** buttons are like mechanical ‘radio buttons’, repeatedly pressing the same button will alternate between that

button and the previously selected button. This means you can repeatedly alternate between **A** & **X** with your eyes closed (or **B** & **X**, or **A** & **B**) because you don't need to move the mouse pointer. Eyes-closed auditioning might improve your listening focus!

As with **A-B** mode, there is access to the input signal (**I/P**) by clicking in the **B** column adjacent to a selected **A** codec. Press **RESET** to reset the internal counters and begin a new test. The display window beneath the **RESET** button displays the number of completed test cycles.

X is randomly set to either **A** or **B**. Listen critically, switching between **A**, **B** and **X**. When you think you have identified **X** as either **A** or **B**, leave the selection on the identified **A** or **B**, and press **CHOOSE**. This will increment the number of test cycles but not display whether your choice was correct. The act of pressing **CHOOSE** also randomly re-allocates either **A** or **B** to **X**. For a thorough and statistically significant test, repeat this audition and **CHOOSE** cycle up to 20 or 25 times.

When you have completed a satisfactory number of cycles, press **REVEAL**, which will show the percentage number of correct guesses:

A 95% confidence level is regarded as statistically significant - anything less than this means you have not reliably been able to distinguish between A and B!

Note:
A 95% confidence level is **not** the same as a 95% correct score! Think of a single cycle; you might by chance guess correctly and obtain a score of 100%, but with a confidence level of only 50%.
As a quick guide, for a 95% confidence level you need a minimum of 8 correct guesses from 10 cycles (80%), 11 from 15 (73%), 14 from 20 (70%), or 17 from 25 (68%).

5.7 Processing Sequence

The concept underlying the development of this plug-in was that it should be the very last item in the mastering chain. A 16 or 24 bit master has been finalised and possibly rendered to a WAV file. What is now required is to take that finalised master and produce an encoded, data-compressed version (or versions) for end-user playback.

In other words, the mix would normally be fed via a limiter to control peaks, and then dither would be applied before the data is truncated and written to the 16 or 24 bit WAV master. Again, under normal circumstances, this WAV file would be passed via an encoder to derive a final MP3 or AAC file for end-user distribution.

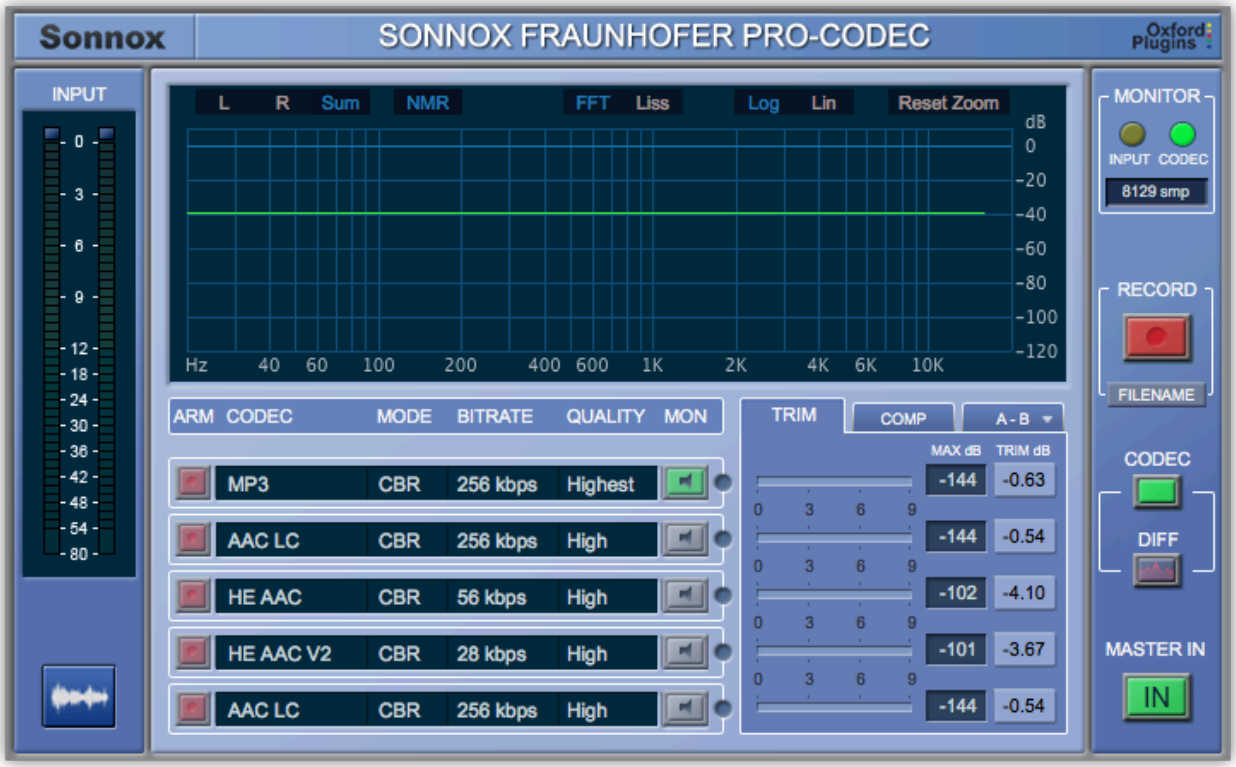
However, please note that this final step is not necessarily carried out currently by the mastering engineer. Frequently, only the WAV file (or AIFF, or lossless encoded file) is sent to the distributor or aggregator. This means that the mastering engineer has no control over which encoders are used, which encoder parameters are used, or whether the final encoded file will clip and/or distort when played back on a consumer device. Also, there is little point in producing a mix optimised for a particular codec or bit rate.

The mastering engineer can now achieve this control by using the Pro-Codec plug-in. Each mix can be auditioned exactly as the end-user will hear it; and consequently the mix can be tuned appropriately by the mastering engineer for a selected codec.

It is important to note that the Pro-Codec plug-in should be used **after** the limiter → dither → truncate process. If a dithered master WAV file is required, it should be rendered with the Plug-In monitor set to input or with the Plug-In bypassed.

5.8 Online Encoding

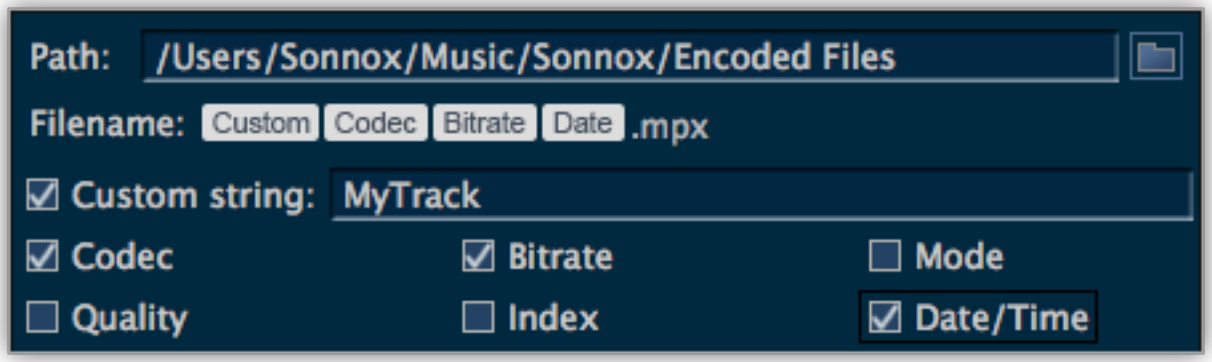
To encode a file in real time, first you have to record enable the codec. This is done by selecting the ARM buttons to the left of the codec list. Multiple files can be selected for simultaneous recording.



To record straight away, press the **RECORD** button when the host is playing back audio; this will write the output of your selected codecs to disk. You cannot change the record **ARM** settings while dropped into **RECORD**:

You can also **ARM** the codec when playback is stopped, so that codecs will start writing the instant you start host playback. Likewise, stopping playback will terminate the writing procedure. Record duration and exact data compression ratios are displayed (see screenshot above).

Press the **FILENAME FORMAT** button to view the filename format pop-over window. This allows the user to specify the destination folder for the recorded files and the output filename elements.



The filename can be composed of elements including codec type, bit rate and date of recording. It is possible to re-order these elements to form a flexible and informative filename. This can be done by clicking on any element within the complete string and

dragging it either to the left or the right to a suitable position. Should you want to add or remove any of these, simply toggle its check box.

Note:
Care should be taken to make filenames distinct, to avoid confusion for files written concurrently. For example, if five mp3 codecs were selected with different bit rates, it would be wise to enable the BIT RATE option for the resultant filenames. The INDEX option is useful to make filenames unique (INDEX indicates the row window number). Furthermore, if repeated renders are likely, the DATE/TIME can be used to distinguish subsequent actions (DATE/TIME includes a field for seconds).

The **Custom String** is restored on a project/session load. When loading Presets, and if you delete the plug-in and re-insert it, the **Custom String** will default to 'My Track'.

The **Output Path** is restored on a project/session load. Otherwise, it remains unaltered, even when loading Presets, and even if you delete the plug-in and re-insert it.

The plug-in always writes the following metadata to MP3 and M4A files:

- Date
- Encoded by manufacturer message ('Sonnox Fraunhofer Pro-Codec Plug-in')

5.9 HD-AAC (Lossless Codec)

A lossless codec is one that produces an output data stream that is an exact duplicate of the input audio data stream. In the simple case of a 24-bit wav file as input, the output of the HD- AAC lossless codec will be a wav file with identical audio data.

When HD-AAC is used in a DAW via the Pro-Codec Plug-In, the situation is slightly more complicated because usually the input is not a simple 16 or 24 bit data stream. Most DAWs use floating point processing, and so do most plug-ins. A floating point datum will have many more states than an audio integer as taken from a 24 bit WAV file (this is another way of saying that the floating point format has higher resolution than the fixed-point wav format). Whenever processing is performed in a DAW or plug-in, the processing is performed at the higher resolution of the floating point system, and the resulting audio data cannot be represented perfectly on a 24 bit scale any more. The data stream will contain floating point values between the quantised values of the 24 bit scale, and these in-between values are called 'inters'. This interactivity can be readily seen using the **DIFF** signal, apparent up to -144 dB for HD-AAC. (One of the main reasons the **DIFF** signal is included is to aid visibility of this situation.)

HD-AAC will not reproduce these inter values as it only uses a 16 or 24 bit scale, and the correct terminology is 'lossless to 24 bits', or 'lossless to 16 bits'.

Therefore, if the audio channel includes any processing (even DAW fades or gain changes), it is important to present correctly truncated data to the input of the lossless encoder. When an audio data stream is truncated, it must first be dithered at the correct level. See section **5.9.3** (Dither and Truncation) for details.

5.9.1 Auditioning HD-AAC

Fraunhofer's HD-AAC has a very clever feature; the single compressed lossless file includes a lossy core channel. It therefore acts as a lossless archival format, a lossless distribution format for the masters, and a final playback format for both lossless and lossy decoders — and all of this in a single file.

During HD-AAC playback, for example, if the decoder in your player has full HD- AAC capability, you will be able to listen to perfect replication of the original wav file. However, the same HD-AAC file will still play through a decoder that doesn't have HD- AAC capability, and instead you will be listening to the lossy embedded AAC channel.

It is possible to audition HD-AAC in real time. If the HD-AAC is selected, and its **MON**

button enabled, the audition point is **after** the HD-AAC encoder-decoder.

You will notice that the **DIFF** signal in the FFT display will drop below -144 dB. (HD-AAC is lossless to 24 bits, so the **DIFF** signal has an amplitude smaller than 24 bits down. If the input signal to the codec is quantised to 24 bits, either by dither and truncation, or because it is unprocessed data from a 24 bit input file, the **DIFF** signal will drop below -180 dB).

While encoding/auditioning an HD-AAC signal, attention should be paid to the lossy core. We recommend selecting the appropriate core codec in an adjacent codec row, which will allow correct auditioning of the lossy codec in the HD-AAC channel, and will also allow correct overload monitoring and mitigation.

The HD-AAC lossy core takes the parameters:

Bit rate

Selectable from 96kbps to 192kbps

Mode

CBR

Quality

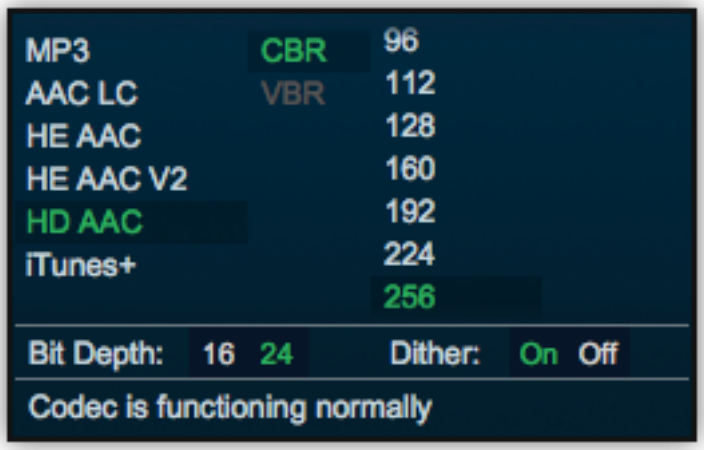
Medium

However, remember that if the Bitstream Level **TRIM** function is used to mitigate the core codec overload, and if that same trim value is applied to the HD-AAC encoder input, then both the lossless and the lossy data streams will be trimmed.

Also note that, under normal circumstances, if the lossless codec's Bitstream Level **Trim** is non-zero to avoid clipping in the lossy chain for an offline encode that uses the online trim values, then the resulting decoded output file will not be lossless because the level has changed.

5.9.2 Bit Depth

The HD-AAC codec has a 16-bit mode that is available in the codec settings selector.



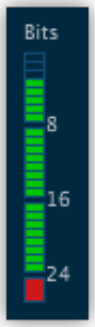
The bit depth and dither settings are reflected in the bit meter operation when moniitoring HD-AAC codec, and they are reflected in the plug-in output signal. The settings are saved into presets and project settings.

5.9.3 Dither and Truncation

When encoding online, or preparing a session for render through the DAW, it is essential to provide the input to the HD-AAC codec with a correctly dithered and truncated audio data stream. As stated above, the plug-in should be the last element of the processing chain, and a dither/truncation plug-in should immediately precede it. This is particularly important if HD-AAC is to be used.

When HD-AAC is selected for audition, the FFT display area changes slightly to include an input bit-meter for the left-hand scale. This bit-meter is included specifically to help select the correct dither/truncation settings for the signal into the codec.

There is one segment per bit for bits 1 to 24 (see right). One large segment at the bottom indicates any activity below the 24th bit.

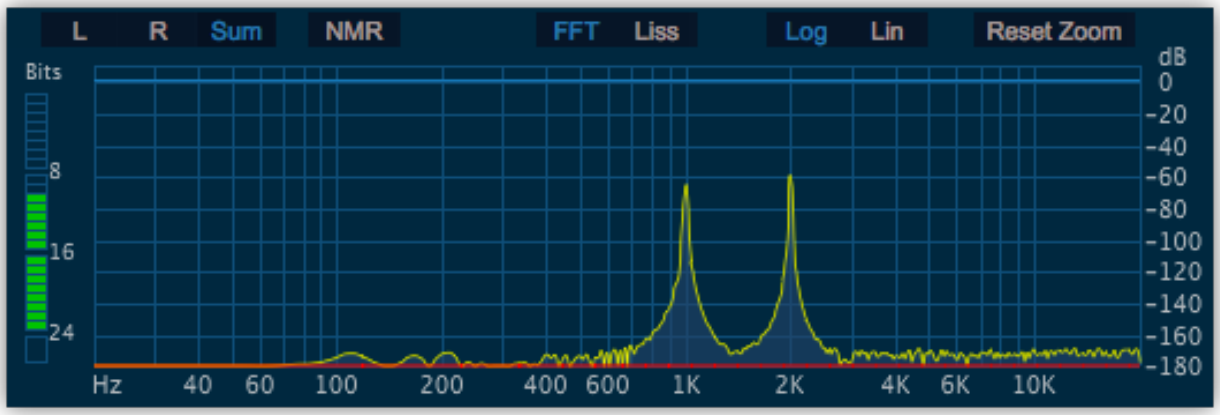


With a 16 bit input you would expect activity in bits 1-16, and with a 24 bit input you would expect activity in bits 1-24. As noted in the introduction to this section, if there is processing before the plug-in, you would expect ‘inters’ and activity on all bits, including bits higher than 24 (even if you start with a 16 bit file).

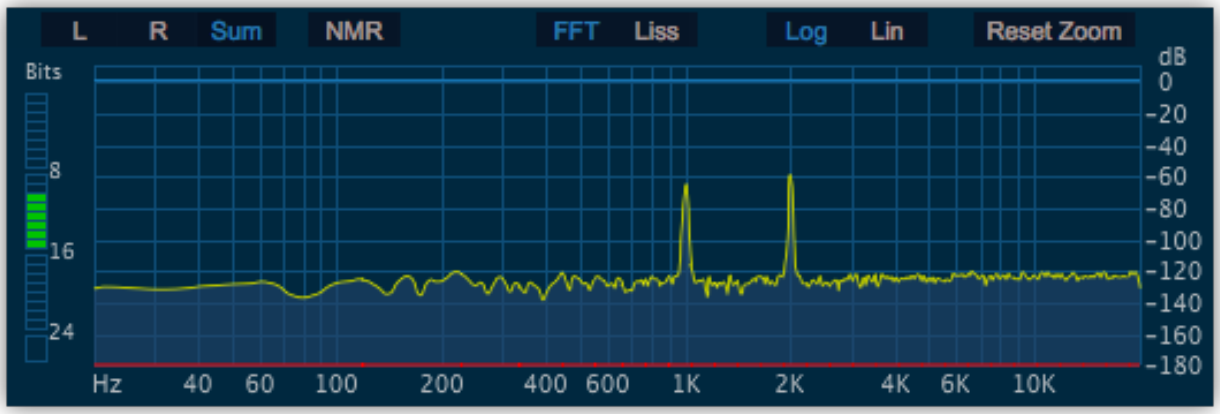
The following images show the plug-in FFT displaying the **INPUT** and **DIFF** signals with varying degrees of dither and truncation.

First, in order to see the effect on the noise floor we input 1kHz and 2kHz tones at -60dB. As this is a low-level signal at -60 dB, there is no activity on the top ten segments of the bit meter.

Note the magnified scale down to -180 dB. (**ALT + drag** down on the dB scale.)



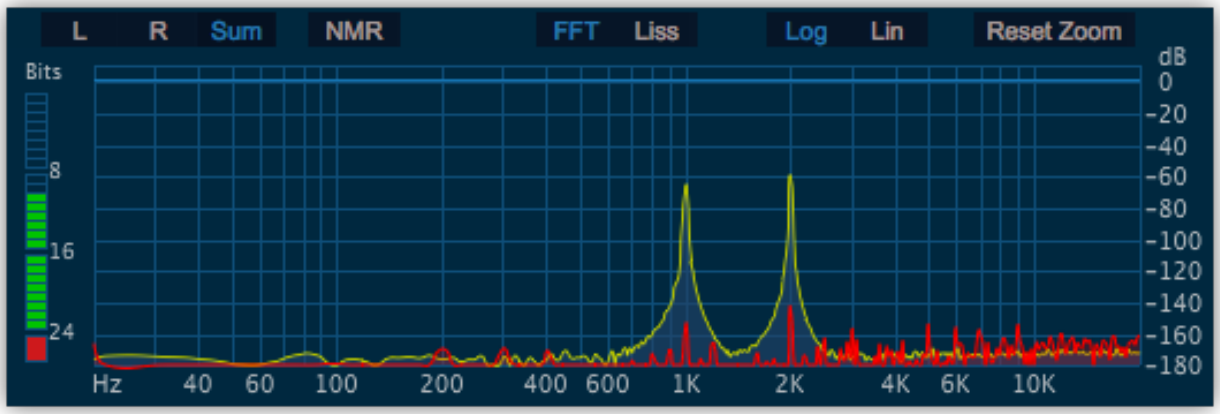
HD-AAC 24 bit with correct dither and truncation prior to the Plug-In.



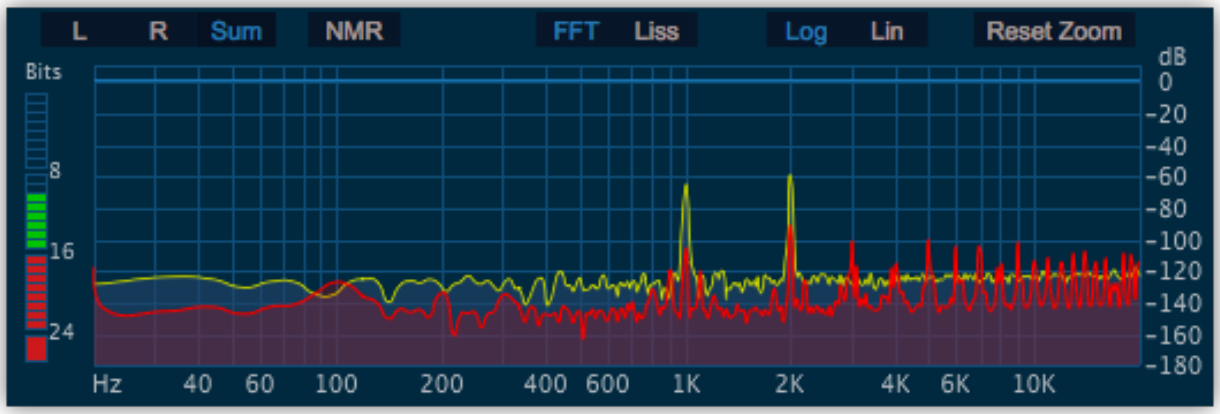
HD-AAC 16 bit with correct dither and truncation prior to the Plug-In.

Since the input signal in both cases has been correctly prepared **prior** to the input of the Pro-Codec Plug-In, no bit activity is shown below the selected bit depth. In these cases, the lossless layer of the encoded HD-AAC file is 100% identical to the input signal.

What if the input signal is not correctly prepared?



HD-AAC 24 bit with **INCORRECT** or **NO** dither and truncation prior to the Plug-In. HD-AAC 24 bit internal dither is **DISABLED**.



HD-AAC 16 bit with **INCORRECT** or **NO** dither and truncation prior to the Plug-In. HD-AAC 16 bit internal dither is **DISABLED**.

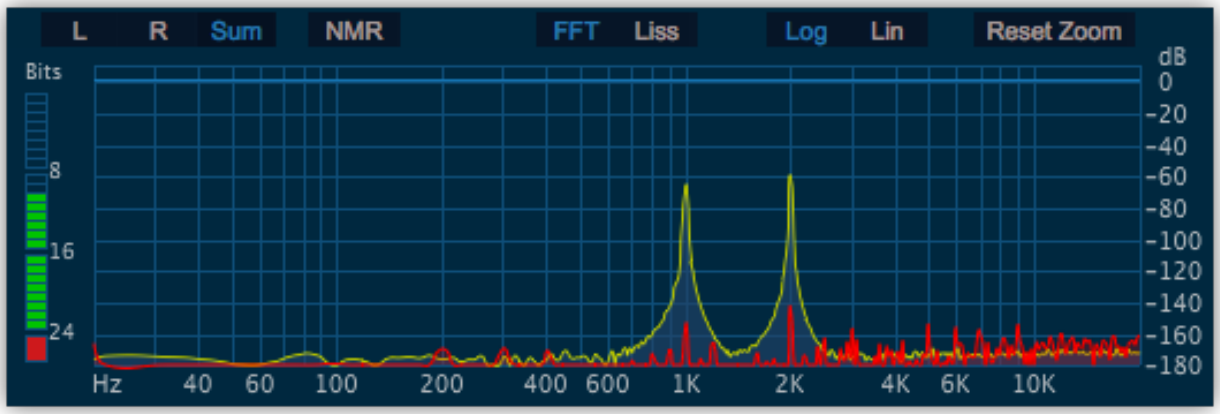
In these cases, it is clear that the HD-AAC truncation to 16/24 bits is causing truncation distortion.

Note:
The HD-AAC processing will still be lossless to 16/24 bits, however, the signal that is losslessly encoded has already been distorted prior to the encoder input, so is not an accurate representation of the signal prior to the Pro-Codec Plug-In!
In these cases, the bottom bits of the bit meter are filled red. This red fill indicates that truncation distortion is occurring!

5.9.4 Internal Dither and Truncation

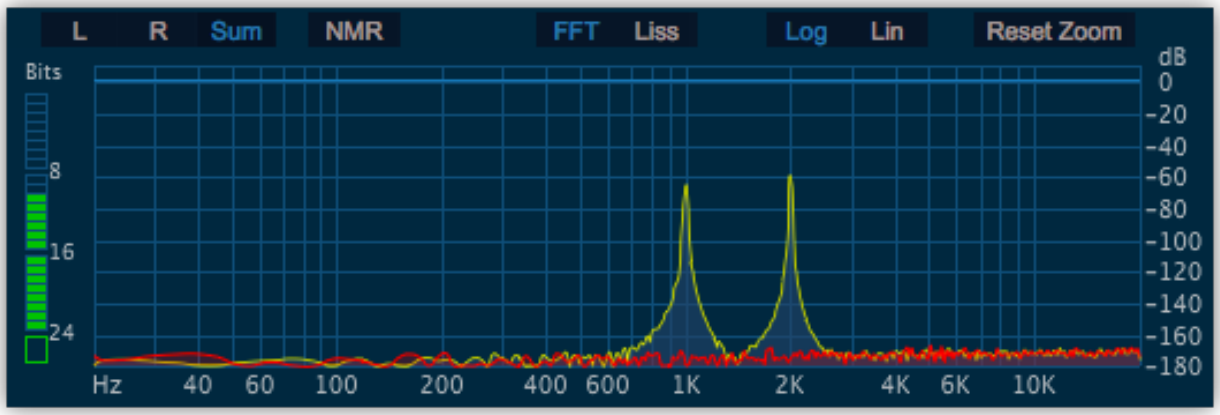
The previous examples include the insertion of a dither/truncation plug-in in the audio path to provide correct treatment of the signal prior to the codec. For convenience, if a dither plug-in is not available, the 16/24 bit dither option can be enabled in the codec settings selector.

> **Note:**
> If the plug-in is subsequently bypassed to render to WAV, the resulting output will not be dithered and truncated correctly.



HD-AAC 24 bit with **INCORRECT** or **NO** dither and truncation prior to the Plug-In. HD-AAC 24 bit internal dither is **DISABLED**.

The **DIFF** signal and the bit meter are indicating presence of audio data input at bits greater than 24. The bits greater than 24 are indicated in red to show an error condition.

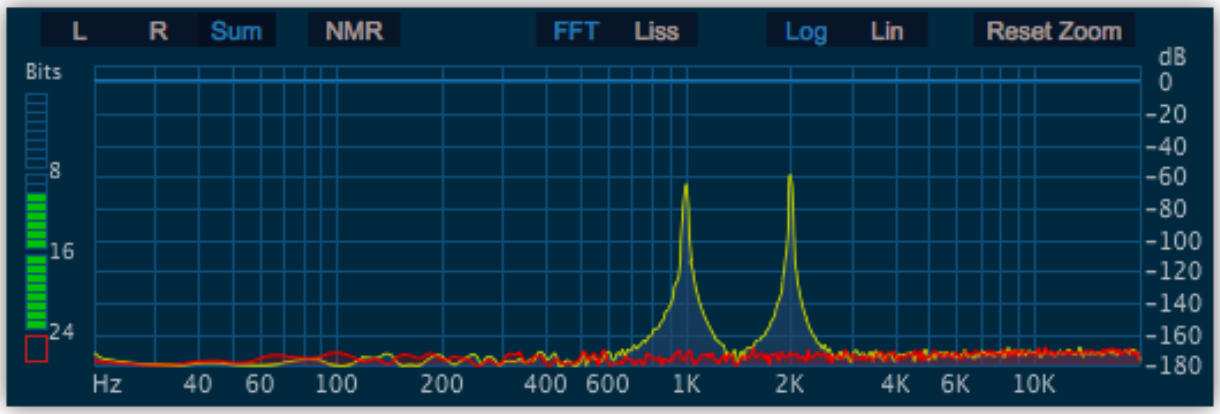


HD-AAC 24 bit with **INCORRECT** or **NO** dither and truncation prior to the Plug-In. HD-AAC 24 bit internal dither is **ENABLED**.

The bit meter indicates that there is no activity in bits greater than 24. The input to the codec is correctly dithered and truncated. The border for the bottom segment is green to indicate that the dither/truncation is provided internally.

The bit meter will also indicate when you are dithering unnecessarily.

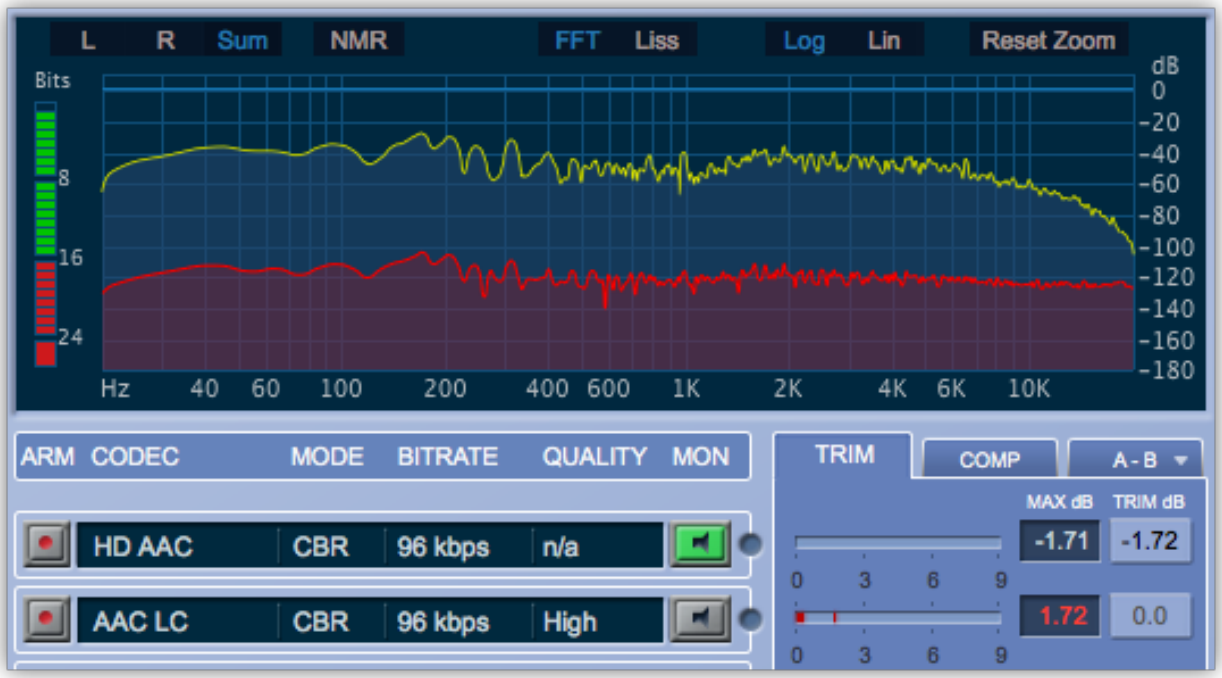
Here a correctly dithered and truncated signal enters the plug-in, but the internal dither is also applied. One of the operations is redundant, and this is indicated by a red border around the bottom segment.



HD-AAC 24 bit with **CORRECT** dither and truncation prior to the Plug-In. HD-AAC 24 bit internal dither is **ENABLED**.

There is a situation when it is desirable to use the internal dither/truncation, even if the signal presented to the plug-in input is correctly treated.

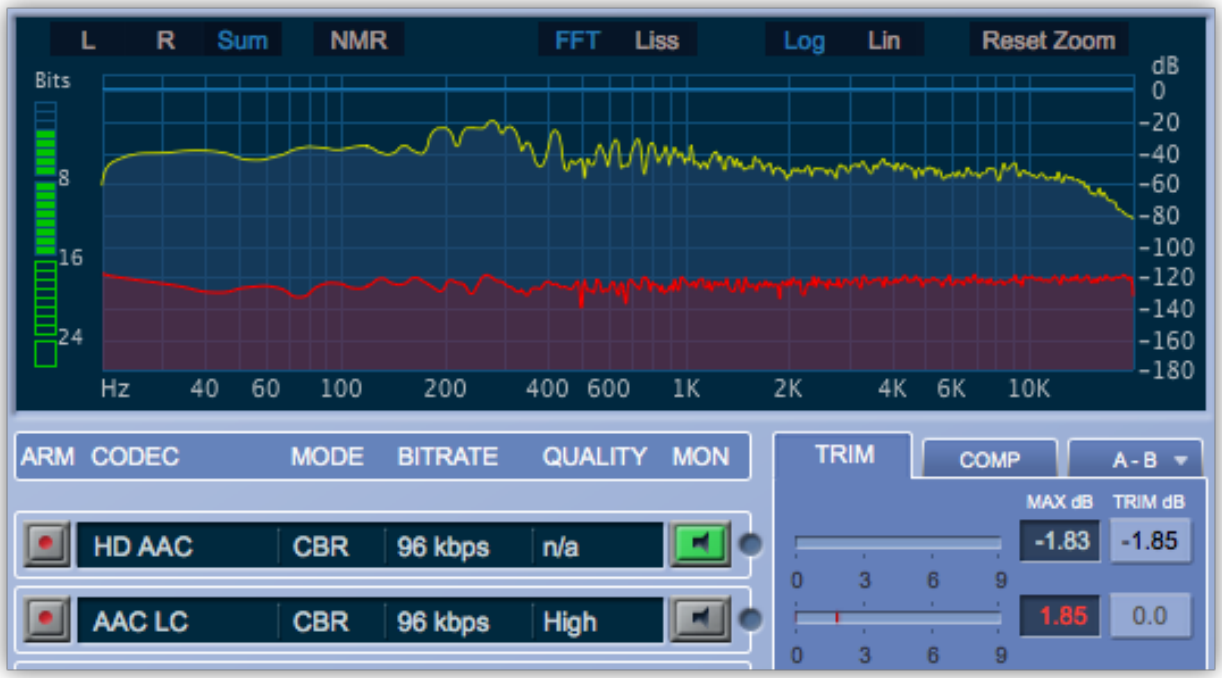
This case is when the lossy core has been observed to cause overloads on decode, and the remedy has been to use the internal **TRIM dB** control. The trim will be applied internally to the plug-in, and any external dithering is not helpful. The signal should be dithered using the internal dither option.



HD-AAC 16 bit with **CORRECT** dither and truncation prior to the Plug-In. Internal Trim is **ENABLED**. HD-AAC 16 bit internal dither is **DISABLED**.

The image above shows a correctly dithered signal input to the plug-in, but the internal trim is non-zero. The bottom segment of the bit meter is flashing red to indicate an error condition.

If the internal dither is enabled, the bit meter indicates that the signal is correctly treated (the bottom segment of the meter is empty) and the active dither is applied internally (indicated by a green border around the segment.

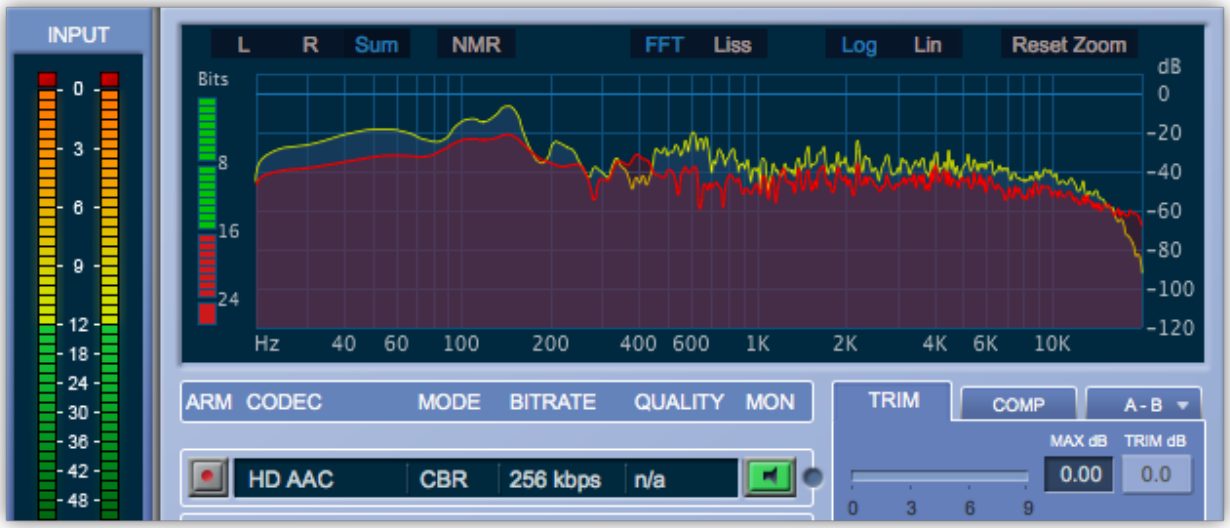


HD-AAC 16 bit with **CORRECT** dither and truncation prior to the Plug-In. Internal Trim is **ENABLED**. HD-AAC 16 bit internal dither is **ENABLED**.

Note:
Again, it is preferable to avoid using the internal trim by lowering the level into the plug-in.

5.9.5 Overloading HD-AAC

The screenshot below illustrates the behaviour of HD-AAC under overloading conditions. A typical audio signal is shown with the display zoomed. The input level is such that the input meters show clipping. Since HD-AAC does not have internal headroom, the signal will clip.



HD-AAC 16 bit with 16 bit input signal overload at the Plug-In input.
HD-AAC 16 bit internal dither is **DISABLED**.

Note the very high **DIFF** signal; the overload condition is clearly audible in the DIFF signal.

5.9.6 Reasons to Audition HD-AAC

Finally, a few observations on the reasons why you should audition a lossless codec, because a hasty person might initially think that it can serve no purpose. If you have carefully read this manual this far, you will have realised there are several good reasons to audition lossless codecs.

Overloading the Input

(See sections 5.3.3 and 5.9.5) The input meters will indicate an overloading input, but the **Bitstream Level** meters will not show clipping for HD-AAC because they have no internal headroom. During audition, the **DIFF** signal clearly shows the effect of overloading the input to the lossless codecs, and the effect will be very audible.

Lossy Core

If it is intended to make use of the lossy core, it is wise to audition the lossy channel

that is buried in the lossless codec. To do this, select AAC-LC CBR with the appropriate bit rate.

Level Change

If a level change is required in the plug-in, and implemented using the **TRIM** tab, again it is wise to verify the audition signal.

Dither and Truncation

It is necessary to correctly dither and truncate the input to the HD-AAC codec (lossless). The audition **DIFF** signal quickly indicates if correct dither and truncation is being applied. Simply put, if any noise is seen in the **DIFF** signal, you have not dithered/truncated the input to the plug-in properly. Note that you might not need to dither/truncate if the input files are playing without any processing; again the **DIFF** signal indicates quickly whether this is true or not.

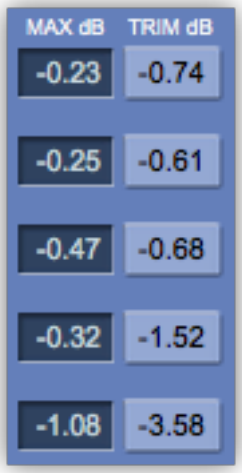
Hearing a Difference

Some people might think they can hear a difference between an input and the output of a lossless codec. Using A-B mode or the A-B-X test, this plug-in allows a user to make up their own mind if they can hear the difference.

5.10 Description of Controls

Touch Pad Controls

Several controls in this plug-in use a ‘touch pad’ associated with a knob or slider. These controls (e.g. **TRIM dB**) give a clear display of the value of a control, and respond to these actions:



Left-Click and drag up/down

Increase/decrease the value

Right-Click and drag up/down

Fine adjustment of value

Double Click

Directly edit value

Shift + Left-Click and drag up/down

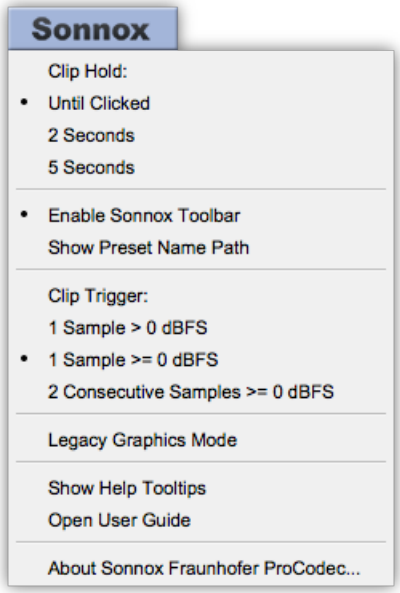
Fine adjustment of value

Control + Left-Click

Set to default

Sonnox Menu Button

This provides access to a number of useful items:



Clip Hold (input meter clip LEDs)

These can be
set to hold until clicked, or for only 2 or 5 seconds.

Enable Sonnox Toolbar

Shows
or hides the Sonnox Preset Manager toolbar.

Show Preset Name Path

Shows or hides the
Preset name path in the Preset Manager toolbar.

Clip trigger

Changes how an overload, or clip is measured on the input meters.

Use Host Channel Configuration

Most VST hosts indicate channel configuration. For those that don't (such as Sound Forge and REAPER), the configuration can be forced to mono or stereo. These options are only visible when the plug-in is inserted on a 6-channel track (or the host does not fully convey the channel count of a mono or stereo track).

> **Note:**
> Some hosts that exhibit this by default provide options to improve the behaviour for such multi-channel-configuration plug-ins. For example, in REAPER 4 and above, right click on the VST Pro-Codec in the FX Browser and enable the option "Inform plug-in when track channel count changes".

Legacy Graphics Mode

This disables the use of OpenGL graphics acceleration for the meters and FFT display, for use with computers with older or incompatible graphics hardware.

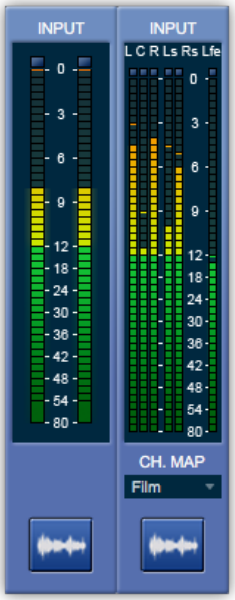
Show help tooltips

This useful tool can be enabled or disabled.

About...

Displays the date, version and build number of the plug-in, and the version of each encoder library in the plug-in.

Input Section



SIGNAL

The input meter indicates 2 dB per LED below -12dBFs, and 0.5dB per LED above -12dBFs. It includes peak-hold indication. The meter will be mono, stereo or multi-channel depending on context.

CH. MAP (5.1 instances only)

Use these options to match the input/output channel mapping of the plug-in to the channel mapping used in your host DAW. If you use SMPTE/ITU channel order, simply select this option.

Manager Button

Click the Manager icon button to open the Manager application for offline encoding, decoding and metadata editing.

FFT Display Section

L/R/Sum/etc

Select which channel's FFT to display.

FFT/LISS

Switch the display between FFT and Lissajous.

NMR

Hide/Show the NMR indicator line on the FFT

Log/Lin

Switch the FFT frequency axis between a logarithmic and linear scale.

Reset Zoom

Reset the zoom level of the FFT frequency and amplitude scales to default.

Codec List Section

CODEC LIST

Displays a list of codecs chosen for auditioning. The codec type, bit-rate and codec mode are displayed at all times. Any codec pre-loaded in the list can be auditioned without glitches. To remove a codec, click the small red ‘x’ at the right hand side of each row.

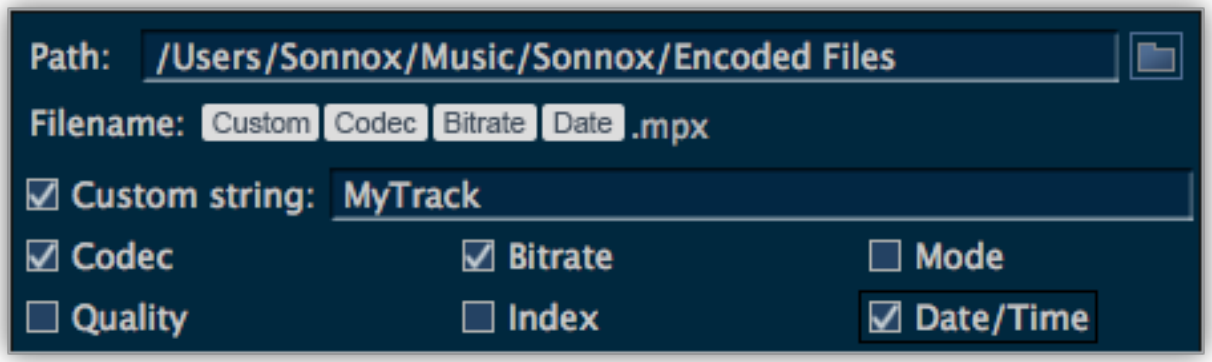
ARM (x5)

Prepares an activated codec in the codec list for writing to disk.

MON (x5)

These select the active codec for routing to the output section. All monitor paths are delay compensated for glitch-free switching.

Filename Format pop-up panel



Path

Shows the destination path for output files. Click the folder browse button to select a new output path.

Filename

Displays the complete string that will be written to an output folder. You can re-order any of the elements for a completely custom filename. The filename elements are:

Custom String

Displays the identifying name string of the filename, eg. a song name.

Codec

Add the codec type. Optional.

Bitrate

Add the encoder bitrate. Optional.

Mode

Add the encoder mode; either CBR or VBR. Optional.

Quality

Add the encoder quality information. Optional.

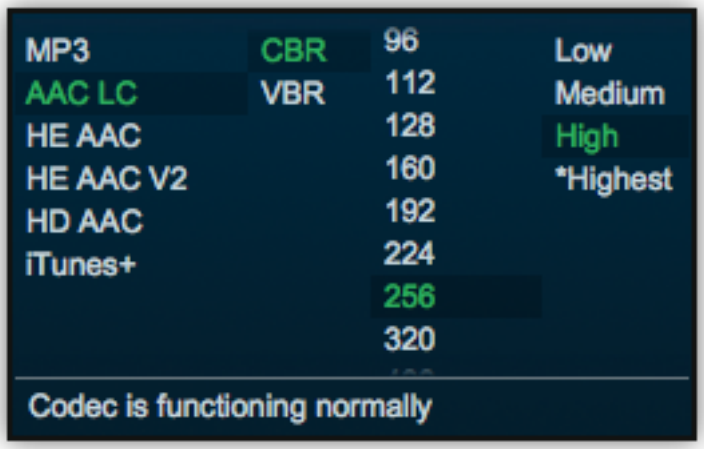
Date/Time

Add the current time and date when the file is created. Optional.

Index

Add a codec specific number index. Optional.

Codec Settings selector



Codec

Click to select a codec from the list. Some codecs will not be available, depending on the host sample rate and the number of channels. For example, HE-AACv2 is stereo only.

Bitrate

Choose the bit rate of the selected codec. Some bit-rates may not be available, depending on the sample rate of the host DAW and the number of channels. If the bit rate is accompanied by an asterisk, the codec is resampling the audio stream.

Mode

Select from the available list either CBR or VBR encoder algorithms for the selected codec.

Quality

Select the quality by which the mode is calculated. The highest quality setting often causes codecs to re-sample the signal (in which case the bit rate and the quality setting will be displayed with an asterisk). If the codec is resampling, the auditioned signal will not reflect this, but the encoded signal written to disk will be at the highest quality setting, which is higher than the auditioned signal. This is the one case where the auditioned signal will not reflect the written signal.

Status

The status (e.g. downsampling) is shown at the bottom of the codec settings selector.

Dither

Only visible if the selected codec is HD-AAC. Only enable 16 or 24 bit dither if the input to the plug-in has not already been dithered and truncated to either 16 or 24 bits.

Downmixing mode

Only visible if the selected codec is AAC-LC MPS or HE-AAC MPS. This allows you to audition the various Fraunhofer downmixing algorithms. By default, no downmixing takes place.

> **Note:**
> When auditioning a 2-channel downmixing mode, the DIFF signal will contain content from the centre, surround and LFE channels, since the DIFF calculation is done using the input and encoded signals.

Output Section

MONITOR

Indicates the plug-in's audio output;
either **INPUT** (ie. the plug-in is bypassed) or **CODEC**. You can switch between input and codec output by clicking on the LEDs. A special case for the mp3-HD codec, whereby the **MONITOR** signal is synthesized, is denoted by an INPUT 16 indication.

TOTAL DELAY

This window shows the total delay through the plug-in; click to toggle between samples or milliseconds. The figure given is the maximum of each individual codec delay. This field is updated in the GUI immediately a new codec or other setting is selected. Depending on the host behaviour, the new value may not be available for delay compensation until the plug-in is removed and re-instantiated.

RECORD

Writes the active codecs pre-selected for record-arm to disk.

Filename Format

This button calls up a pop-up used to configure the encoded file data.

CODEC

When selected, will route the encoded-decoded signal to the output.

DIFF

When selected will route the difference signal to the output. The difference signal is the encoded-decoded signal subtracted from the unprocessed input signal.

MASTER IN

With the **IN** button selected, all processing is active. If not selected, the outputs are disabled. However, much of the processing remains active, for example, the FFT display and the overload detection. The **MASTER IN** control is automatable.

The five **MON** buttons and the **MASTER IN**, **CODEC** and **DIFF** buttons all provide smoothly interpolated and glitch-free audio switching. It is advisable to use these for auditioning, rather than using the host application's bypass function (which is commonly not interpolated).

Bitstream Levels / Trim Tab

METERS (x5)

Indicate when a decoded signal is above 0 dB. Includes peak hold.

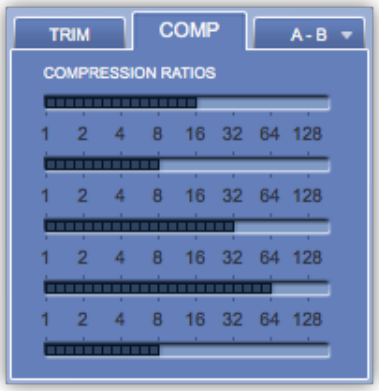
MAX dB (x5)

Indicates the maximum level detected by the decoder. This value is white if negative, and red if positive. Clicking in this window resets the peak hold value.

TRIM (x5)

Indicates the input trim to the individual codecs. This is a Touch Pad control, allowing a fader style movement to determine an attenuation of up to a maximum of 12 dB. If this value is non-zero, the trim value is displayed darker. The TRIM dB controls can be automated.

COMP (compression) Tab



COMPRESSION RATIOS (x5)

Shows an indication of the amount of data compression in the output data stream or file. The compression is expressed as a ratio, from 1:1 down to around 128:1. A yellow asterisk denotes an estimated value.

A-B Auditioning Tab[1]

A (x5)

Selects the codec to be routed through **A**.

B (x5)

Selects the codec or input to be routed through **B**.

A/B Glitchlessly switches output routing between active codecs selected in **A** or **B**.

HIDE

Cloaks the AB matrix state and FFT scope.

BOTH

Displays both **A** and **B** difference signals.

A-B-X Test Tab[1]

A (x5)

Selects the codec to be routed through **A**.

B (x5)

Selects
the codec or input to be routed through **B**.

A/B/X

Glitchlessly switches output routing between active codecs selected in **A** or **B**, or the anonymous codec in **X**.

RESET

Sets the ABX machine back to a start condition. Resets counters to zero and sets **X** to either **A** or **B**.

CHOOSE

Increments cycle counter. Examines whether choice of **A** or **B** correctly matches **X**. If correct guess, increments the success counter. Randomly re-allocates **A** or **B** to **X**.

REVEAL

Calculates the 'success' counter divided by the cycle counter and displays the result.

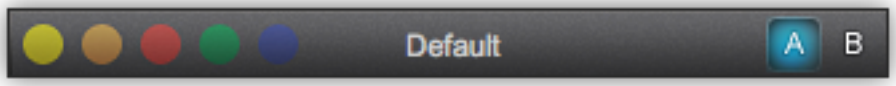
DISPLAY

Shows number of cycles completed during test; when **REVEAL** is pressed, it will show the success ratio.

Note:

[1] Select the **A-B** Auditioning Tab or the **A-B-X** Test Tab via the Sonnox menu button

6 Preset Manager Toolbar



The Sonnox Fraunhofer Pro-Codec Plug-In comes equipped with its own onboard Preset Manager, which is displayed at the top of the plug-in window. The reasoning behind this is to allow increased portability of your presets across all the host applications, while also providing a consistent and versatile interface. While most host platforms allow creation and loading of presets, those host-created preset files are not portable between different host applications. With the Oxford plug-ins' Preset Manager, you can create a named preset in one host application and load it when using an alternative application.

The Sonnox Preset Manager is fully described in a companion document — **Sonnox Toolbar and Preset Manager User Guide** — available for download at www.sonnox.com/docs

6.1 Presets and Project/Session Data Handling

The Sonnox Fraunhofer Pro-Codec plug-in contains internal data not normally found in a plug-in, such as filenames and folder paths. This has prompted the adoption of a slightly different preset model in order to allow the creation and utilization of independent libraries of data presets.

6.1.1 Internal Data Rules

The following special data have special rules applied to them:

- The Export Settings data (the Custom String, filename attributes and Output Path)
- The **Master Record** button and five Record **ARM** buttons

Unless detailed otherwise, all other data are always returned in any form of preset load.

Rule 1: Project / Session loads (and host preset loads)

Will load all data as it was saved, including all the Special data above.

Rule 2: Onboard Preset Manager (PM) preset loads (and PM copy/pastes, +/-)

Will not load any of the special data listed above.

To be more specific:

Output folder
Remains as set.

Output filename elements
Remains as set.

Custom String
Reset to the default ‘MyTrack’.

TRIM dB
All five are reset to 0 dB.

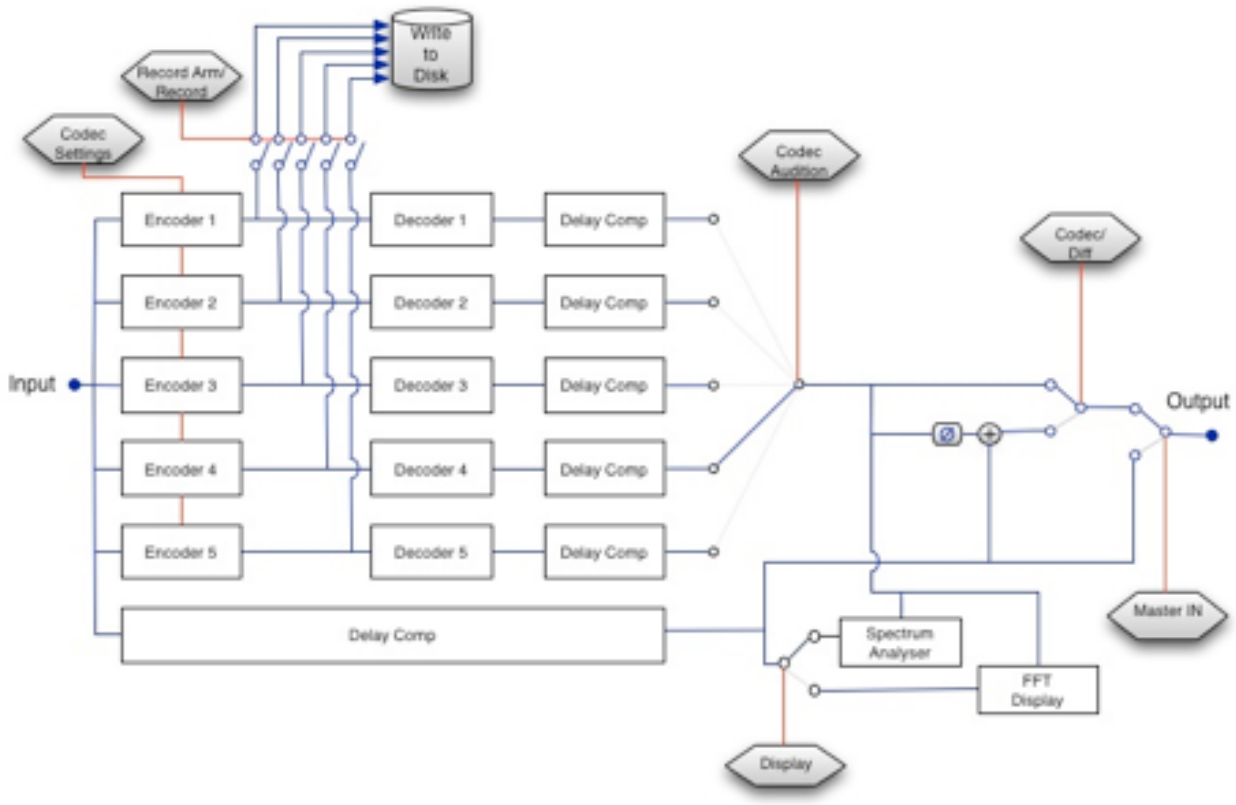
Record ARM
Master Record and all five codec record arm buttons are reset to off.

Note:
Sonnox Toolbar preset loads must match the session sample rate and the session channel configuration to avoid incompatible codec settings (the codecs are sensitive to session sample rate and channel configuration). Presets that are incompatible with the current instance will appear greyed out in the preset load menu.

Rule 3: Persistence for plug-in insertion and deletion:

Only the **Output Path** and filename **Attributes** are remembered across a delete/insert. All other controls (whether special or otherwise) are not remembered, and so are defaulted on plug-in insert (notably the Custom String is defaulted to 'MyTrack').

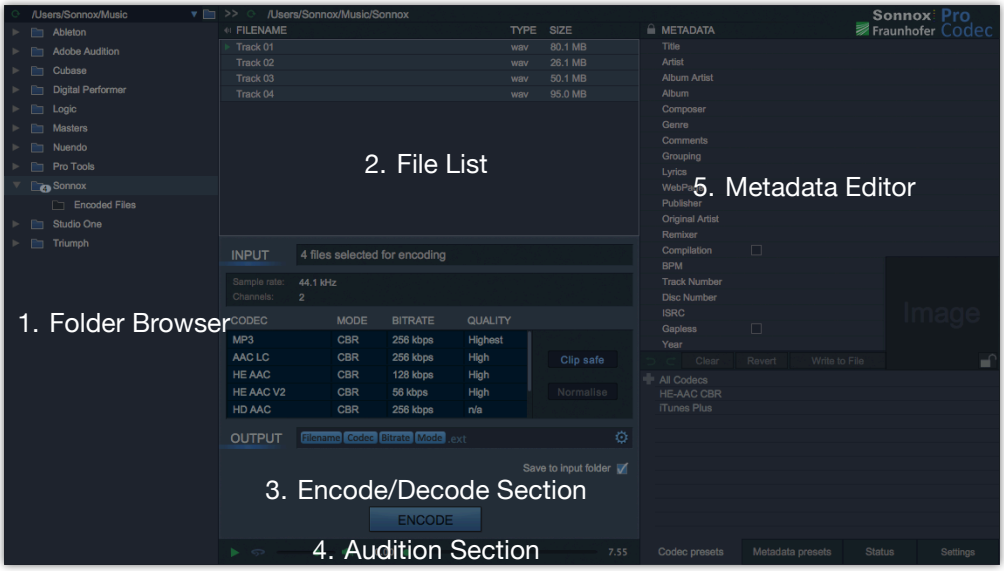
6.2 Plug-In Signal and Control Flow Diagram



7 The Pro-Codec Manager Application

The Pro-Codec Toolbox Manager is a standalone application primarily intended for encoding files using the latest codec technology from Fraunhofer IIS. In addition, the Manager provides the ability to decode files, and features a metadata editor for adding metadata to your encoded files.

The user interface of the Manager is separated into five main sections:



1. The **Folder Browser** is used to navigate folders on your file system containing supported media. Selecting a folder containing files supported by the manager will load them into the File List. The selected folder is denoted by an orange highlight.
2. To the right of the Folder Browser is the **File List**. It provides a detailed view of all supported files in the folder currently selected by the Folder Browser.
3. Below the File List is the **Encode/Decode Section**. This is where settings pertaining to encoding and decoding can be adjusted, as well as output file name and location options.
4. The **Audition Section** is situated at the bottom of the window. It contains controls for playing back audio files selected for audition in the File List.
5. Occupying the top right hand area of the screen is the **Metadata Editor**. If a supported M4A or MP3 file is selected, any metadata it contains can be examined, edited, and written back to the file. If a WAV or AIFF file is selected, the section is used to edit the metadata that will be written when this file is encoded.

7.1 Folder Browser

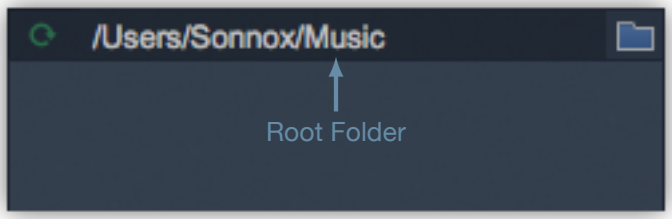
The leftmost section of the Pro-Codec Manager presents controls for browsing folders on your system.

From here, individual folders can be selected so that their contents can be loaded into the File List. This section describes the different components that make up the **Folder Browser** section, including details about some common operations you may need to perform.

To help minimise the size of the Manager, the **Folder Browser** can be hidden and revealed at any time using the hide/reveal button. This button is located on the top right hand side of the section.

7.1.1 Setting up a Root Folder

The Root of the **Folder Browser** is the folder whose path is displayed at the top of the section. All folders contained in the current Root Folder will appear in the **Folder Browser**.



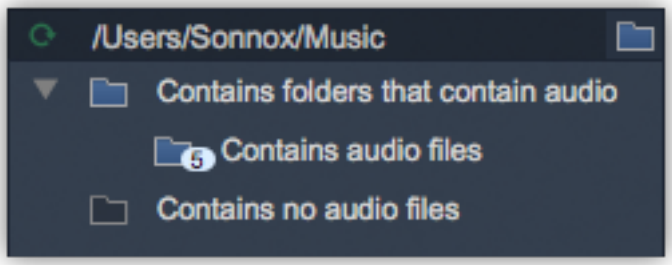
The Root Folder can be selected in one of five ways:

- Select one of the default Root Folders from the drop-down list.
- Browse to a new Root Folder via the folder browse button.
- Right-click a folder in the Folder Browser and choose **Make Root**.
- Drag and drop a folder onto the section. This folder will become the Root.
- Drag and drop a supported audio file onto the section. This file's parent folder will become the Root.

7.1.2 Finding Audio Files on your System

The **Folder Browser** includes a feature designed to help locate audio files on your system. Thus, using the built-in **Folder Browser** may be more efficient than browsing to folders using the operating-system pop-up browser (accessible via the **Browse** button at the top of the section). This feature works as follows:

- Folders that contain audio files in any of their sub-folders are displayed with a full-colour blue folder icon.
- Folders that contain audio files in their top level are displayed in full colour along with the number of audio files that they contain.
- Folders that do not contain audio files in their top level, or any of their sub-folders are displayed with an empty folder icon.



This feature can be CPU intensive. If the root of the **Folder Browser** contains a significant number of sub-folders, or is located on a slow external storage device, the background scanning of all folders in the **Folder Browser's** tree can be correspondingly slow.

If this becomes a hindrance on your system, the background scan can be disabled in the Settings tab, by deselecting **Folder Browser: Background scan for audio files**.

With this feature disabled, all folders will be displayed with a full-colour folder icon, and the number of files in each folder will not be shown.

7.1.3 Supported File Types

The Manager supports the following file types:

- M4A AAC (including AAC-LC, HE-AAC and MPEG Surround)
- MP3
- WAVE (Uncompressed)
- AIFF (Uncompressed)

Unsupported file types will not be recognised as audio files by the background scan process, nor will they be displayed in the File List.

7.2 File List

This section displays all compatible audio files that are in the top level of the folder currently selected in the Folder Browser. Incompatible file types are not displayed. The section provides the following information:

- File Name
- File Type
- File Size (in MB)
- Presence of Metadata (for MP3 and M4A files). If a file contains metadata supported by the Manager, an Arrow Button is displayed in the rightmost column of the list.
  - **Hover** the mouse cursor over one of these buttons to preview the file's metadata.
  - **Single-click** to import the file's metadata into the Metadata Editor.

To select a file to audition, **single-click** in the playback column to the left of the file name. The file will begin to play immediately, and clicking in the same column again will pause playback.

To select a file for processing, **single-click** on the file.

7.2.1 Selecting Multiple Files for Batch Processing

It is possible to select multiple files in the **File List**, for encoding/decoding to the same output format. To select multiple files:

- **Command+Click** (Mac) or **Control+Click** (Windows) to add a file to the current selection.
- **Shift+Click** to select a range of files.
- Press **Command+A** (Mac) or **Control+A** (Windows) to select all.

It is important to remember that the available bit rate list for each codec changes depending on the source sample rate and channel configuration.

Because of this, and the modality of the **Codec Selector** and decoder output format selector, there are necessary restrictions on the types of files that can be selected simultaneously:

1. Encoding and Decoding cannot coexist in a single batch operation
2. When selecting uncompressed WAV or AIFF files for encoding, all files in the selection must have:
    - the same sample rate
    - the same channel configuration

In order to enforce these restrictions, when keyboard focus is set to the File List, and a multi-select modifier (**Command** (Mac), **Control** (Windows), or shift) is held, incompatible files in the list are greyed out and cannot be selected.

Drag an audio file onto the **File List** to set the file's parent folder as the root of the **Folder Browser** and select the file for processing. Please note that this will only work for audio file types that are supported by the Manager, such as WAVE, AIFF, MP3 and AAC encoded M4A.

Right-click a file in the list and select **Reveal in Finder** (Mac) or **Reveal in Explorer** (Windows) to open the operating system file browser at that file's location.

7.3 Encode-Decode Section

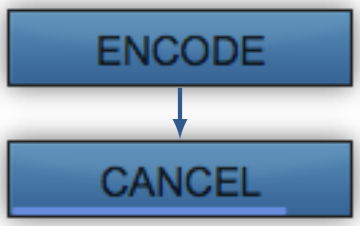
7.3.1 Encoding

The **Clip Safe** button enables a level-correction feature. This works by decoding the newly compressed file, calculating the maximum sample value in the PCM stream and, if necessary, applying the precise amount of gain trim required when the source file is encoded a second time. This guarantees that the final encoded file, once decoded for playback, will not contain any illegal sample values (or overloads). However, if your audio files have been correctly prepared using the Pro-Codec Plug-in, using **Clip Safe** should not be necessary.

NORMALISE can be enabled in addition to **Clip Safe** in order to ensure that the post-decoder level of the encoded file is as close to 0 dBFS as possible.

If the **Normalise** button is enabled, the source file is encoded, then decoded and the maximum sample value is found. The source file is then re-encoded with a gain adjustment to force the decoded max value up or down to 0 dB.

When you are ready, press the **Encode** button to encode the file and write it to the target location. A status bar will indicate progress during the encoding process. During encoding, it is possible to press the encode button again if you wish to cancel the process.



Note:
The current status of the process is also indicated in the **Status** tab at the bottom right corner of the Manager.

All edited tags in the **Metadata Editor** will be written to the encoded output file.

Some computers may indicate a ‘spinning progress cursor’ during the first few seconds of the encoding process. This is quite normal and does not indicate that the application is not responding.

7.3.2 Codec Settings Presets

Codec settings can be saved as presets for later use. To display the codec settings preset manager, click the **Codec Presets** tab in the lower right-hand area of the Manager.

Hover the mouse cursor over a preset name to preview the saved codec settings. Click on a preset name to load these settings into the **Codec List**.

The list of presets is ordered alphabetically, and is immediately re-ordered when a new preset is saved.

To delete or remove a preset, hover the mouse cursor over the preset name, and press the red **X** button that appears at the right-hand side of the row. Since this is not undoable, a pop-up window will request confirmation before deleting the preset.

If you have saved a Pro-Codec Plug-In preset of various codec settings that you wish to use, you can load this into the Manager application as above (and vice versa). By default, any presets saved by the Plug-In will appear in the **Codec Presets** tab in the Manager application, and vice versa.

Note:
The Plug-In has two separate banks of settings. Only the codec settings from the active bank of a Plug-In preset will be loaded into the Manager application. The active bank can be either **A** or **B** – whichever bank was visible (in use) when the preset was saved! See section **6** (Preset Manager Toolbar) for details).

7.3.3 Decoding

The Manager can also decode compressed files for import into a host sequencer (or for general purpose decoding). To decode supported M4A or MP3 files, simply select them in the **File List**, choose WAV or AIFF, and press the **DECODE** button.

The **CLIP SAFE** button enables a level-correction feature which is essentially the same as **CLIP SAFE** when Encoding. It works by decoding the compressed file, calculating the maximum sample value in the PCM stream and, if necessary, applying the precise amount of gain trim required to ensure that the peak level is less than 0 dBFS, before writing the output file to disk at your chosen bit depth.

Files that were previously encoded without correct attention to the post-decoder peak level may have caused clipping in the DAC during playback. These files can now be

decoded to WAV/AIFF with the peaks intact and reduced to below 0 dBFS, potentially restoring some fidelity to the programme material.

If the source audio file is very quiet, **NORMALISE** can be enabled in addition to **CLIP SAFE** in order to ensure that the decoded level of the file is as close to 0 dBFS as possible.

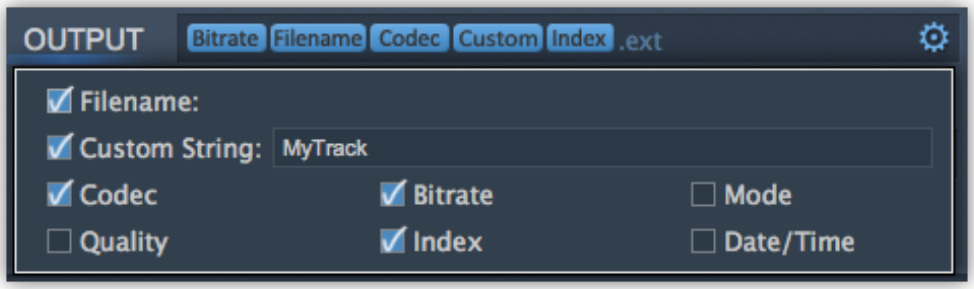
Use the **Output Path** field to select where you would like the decoded files to be saved to. By selecting the **save to input folder** option, the source folder will be used. When **save to input folder** is disabled, a further output folder field is displayed. Click the browser icon on the right to choose your output folder. Click the output filename to rename the destination file.

7.3.4 Batch Encoding/Decoding

It is possible to select multiple files in the File List for encoding, decoding and metadata editing.

Each file in a selection will be processed in the same way. When encoding and decoding, each file will be encoded or decoded to the same output format. When writing metadata (via the **Write to File** button), all tags in the Metadata Editor will be written to each selected file. The only exception to this are the few tags that may not be supported by the filetype. See section 7.5.3 (Supported Tags) for details). Unsupported tags will simply be ignored for that file.

When batch encoding, use the **Filename Format** attributes to avoid filename clashes.



Click the cog to open the Filename Format panel

7.3.5 Dither

When **Clip Safe** or **Clip Safe + Normalise** are used, the Manager may need to dither after applying the required gain trim.

When **encoding**, dither is only applied if an HD-AAC encode produces overs and requires some gain trim before the re-encode process. When **decoding**, dither is applied when any encoded file (HD-AAC or otherwise) requires gain trim before being written to 16 or 24 bit WAV/AIFF.

7.4 Auditioning Files in the Manager

It is possible to listen to any supported audio file using the Pro-Codec Manager. Click in the audition column to the left of the file name to load a file into the internal playback section. The file will begin to play immediately. Press the spacebar to pause or resume playback. Use the **SETTINGS** tab to select your audio output device.

This section provides familiar controls – play/pause, volume and play position seeking. Clicking the speaker icon toggles playback mute. The loop button has three modes:

- No looping
- Loop the currently selected file
- Loop all files in the list in sequence. At the end of playback, the next file in the list is selected, and playback begins. Playback will stop at the end of the last file in the list.

The audio output device used by the Manager can be selected from the **SETTINGS** tab. If a device is connected to the system while the Manager is open, press the rescan button, and the device will become available via the drop-down list.

Note:
It is recommended to configure the plug-in's host DAW to release the audio driver in the background. This will prevent the device driver from attempting to change your session/project sample rate, which can cause problems with some host DAWs.

7.5 The Metadata Editor

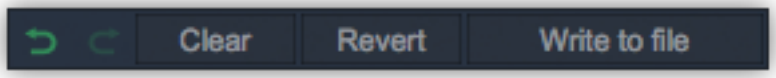
7.5.1 Overview

The MP3 and M4A audio file formats can optionally contain information about a file. This information is commonly known as **METADATA**. Typical metadata tags could include the Title, Artist, Track Number, and Cover Art. The Title tag of a file is distinct from its file name.

Adding metadata to your audio files makes them easier to identify, and allows other applications to categorise and sort them based on your needs. It is also a sign of quality, offering the information that you would normally find on an album CD case or vinyl sleeve.

The Pro-Codec Manager provides the ability to add the most commonly used tags to your files during the encoding process, and edit tags in existing MP3 and supported M4A files.

When an MP3 or M4A file is selected in the **File List** (single-click), any supported metadata it contains is imported into the **Metadata Editor**. Once a tag has been edited, its row is displayed with a light grey highlight. Metadata in files that are not currently selected can be previewed in context, by hovering the mouse cursor over the Metadata Import button in the **File List**.



Metadata section buttons

Undo/Redo

Undo or Redo a metadata edit. Importing metadata from presets or other files sets an undo point so can also be undone/redone.

Clear

Remove all the existing metadata in the table.

Revert

Undo all changes and restore the section to the metadata held within the selected file.

Write to File

Write the contents of the Metadata Editor to the selected file. **This alters the selected file, and cannot be undone**.

7.5.2 Supported Tag Formats

The Manager provides the ability to add and modify metadata in files with an ‘mp3’ extension, and some files with an ‘m4a’ extension. Adding and modifying metadata in m4a files compressed with the Apple Lossless Audio Codec (ALAC) is not supported. The Status panel will display an error message if one of these files is selected. The adding and editing of metadata in ‘wav’ and ‘aiff’ formats is not supported.

The metadata contained within mp3 and m4a files is distinct from the audio data, and is stored in a separate format of its own. The Manager supports the handling of 2 different metadata formats:

ID3v2

This format is typically for use in mp3 files. The ID3v2 standard is now in its 4th revision, but many players, including Windows 7 utilities, only support up to ID3v2.3. The Manager can read and write metadata in all versions: 2.2, 2.3 and 2.4.

iTunes Metadata

This is the standard developed and popularised by Apple, and found in m4a files.

These two tag formats are widely supported, and can be parsed by most commercially available music player applications.

Note:
M4A files can contain metadata in the ID3 format, but MP3 files cannot contain metadata in the iTunes format.

7.5.3 Supported Tags

The list of available tags varies between MP3 and AAC files due to the different metadata specifications of each format. Tags that are not supported for the selected file type will be disabled, and appear in italics.

TAG	MP3 Files	AAC Files	More Info
Title	✓	✓	
Artist	✓	✓	
Album Artist	✓	✓	
Album	✓	✓	
Composer	✓	✓	
Genre	✓	✓	Standard or nonstandard
Comments	✓	✓	Multi-line text
Grouping	✓	✓	
Lyrics	✓	✓	Multi-line text
Webpage	✓	not supported	
Publisher	✓	not supported	
Original Artist	✓	not supported	
Remixer	✓	not supported	
Compilation	✓	✓	Toggle
BPM	✓	✓	Numeric (three digits)
Track Number	✓	✓	Numeric (e.g. 1/12)
Disc Number	✓	✓	Numeric (e.g. 1/12)
ISRC	✓	✓	
Gapless	✓	✓	
Date	✓	✓	
Cover Art	✓	✓	

7.5.4 Editing Text Tags

In order to edit a text tag, click on the tag row in the **Metadata Editor**. This opens an inline text field; type your changes into this field.

There are two kind of text tag:

- Single line
- Multi-line (Lyrics and Comments)

To accept changes for a single line tag, press the **Enter** key, or press **TAB** or **Shift + TAB** to accept and move to the next or previous tag.

In multi-line tags, pressing **Enter** will insert a new line into the tag text field.

To accept changes for a multi-line tag, press **Command + Enter** (Mac) or **Control + Enter** (Windows). Alternately, press **TAB** or **Shift + TAB** to accept and move to the next or previous tag.

The **Enter** key has one more behaviour in the Metadata Editor. If there are no tags currently being edited, press **Enter** to move back to and edit the most recently touched tag. This can be useful if you have just accepted an incorrect edit. Rather than clicking on the tag again, or tabbing back/forwards, simply press the **Enter** key.

While editing a text tag, press **Escape** to discard your edits.

Some fields have format restrictions and will only accept a certain type and/or number of characters. See the Supported Tags table for details.

7.5.5 Adding Cover Art

To add Cover Art to your encoded files, first select a file by **single-clicking** it in the **File List**.

There are two ways to load an image file into the Metadata Editor:

- Click on the Image tag field in the lower right corner of the **Metadata Editor**, and browse to an image file.
- **Drag** an image file onto the Manager

To remove an image file from the **Metadata Editor**, hover your mouse cursor over the **Image** tag, and click the red **X** button that appears. If this image file exists in the **Cover Art** tag of the selected audio file, press the **Write to File** button to overwrite the previous **Cover Art** tag with empty contents.

7.5.6 Writing Tags to Multiple Files

Metadata tags are written to all selected files. Unsupported tags will simply be skipped. For example, if M4A and MP3 files are selected, the Gapless tag will be skipped for all MP3 files, but still written to the selected M4A files.

7.5.7 Importing Tags from Another File

Any audio files that contain metadata are listed in the File List with a small arrow button in the right-most column.

Hover your mouse cursor over a button to **preview** the metadata from that file in the Metadata Editor. Previewed metadata tags are shown in green. Click a button to **import** the metadata from that file into the Metadata Editor.

Only non-empty tags will be imported, so existing edited tags cannot be overwritten with an empty tag from another file.

Importing metadata from another file in this way is compatible with the **Undo/Redo** mechanism, and the **Revert** feature.

7.5.8 Metadata Presets

Metadata tags can be saved into presets for later use. To display the metadata presets manager, click the **Metadata Presets** tab in the lower right hand area of the Manager. In this presets manager, a list of saved presets is displayed.

Hover the mouse cursor over a preset name to **preview** the saved tags in the Metadata Editor. Previewed metadata tags are shown in green. Click on a preset to **load** the saved tags into the Metadata Editor as changes for the selected file.

Since the presets are imported as changes to the existing content in the **Metadata Editor**, if a preset does not contain anything different, nothing will be previewed or imported.

To undo metadata tag changes after importing a preset, press the **Undo** button, or use the keyboard shortcut **Command + Z** (Mac) or **Control + Z** (Windows).

To save a new metadata preset, press **Command + S** (Mac) or **Control + S** (Windows), or press the **+** button that appears on mouse over in the upper-left corner of the Preset Tab. This will open a text field where the preset name can be typed. Press **Enter** to save the preset, or **Escape** to dismiss the text field, and cancel the saving process.

The list of presets is ordered alphabetically, and is immediately re-ordered when a new preset is saved. Empty presets cannot be saved, so the **+** button is not displayed when the **Metadata Editor** is empty.

To delete or remove a preset, hover your mouse cursor over the preset name, and press the red **X** button that appears at the right-hand side of the row. A pop-up window will ask for confirmation before deleting the preset.

7.5.9 The Metadata Lock Mechanism

It is important to remember that the changes performed in the **Metadata Editor** are not written to the selected file until the **Write to File** button is pressed.

If a new file is selected for processing when the metadata of the previously selected file has been edited (but not written to file), those changes will be lost. A pop-up window will appear in this case, to warn the user and allow the operation to be cancelled.

Each field in the **Metadata Editor** can be protected to ensure that it isn't modified during editing. Clicking at the beginning of that field's row, in the padlock column, will toggle a small padlock on that row.

Click and drag in the padlock column to lock or unlock multiple tags.

Locked rows will retain the current tag content when clearing, reverting, or importing metadata from other files or presets. Additionally, when changing the selected-for-processing file, that row will keep its state. The **Cover Art** tag can also be locked, by clicking on the padlock icon in the lower right corner.

When a non-encoded file is selected for processing, metadata info can also be added in the **Metadata Editor**. The main difference with encoded files is that the **Write to File** button is disabled, since this metadata cannot be written to an uncompressed PCM file. Instead, the metadata in the section will be written to the new encoded file.

7.6 Keyboard Shortcuts

Global Context	Mac	Windows
Process File(s)	Command+P	Control+P
Write metadata	Command+W	Control+W
Undo metadata edit	Command+Z	Control+Z
Redo metadata edit	Shift+Cmd+Z	Shift+Ctrl+Z
Save metadata preset	Command+S	Control+S
Toggle playback	Spacebar	Spacebar

The grey highlight that appears around the **Folder Browser**, **File List** or **Metadata Editor** indicates the section that is currently the keyboard focus target for the remaining key commands. Focus can be changed by pressing **Tab** or **Shift + Tab**, and also follows the most recently clicked section.

Folder Browser	Key Command
Select next or previous folder	Arrow key Up/Down
Collapse folder	Arrow key left

File List	
Select next or previous file	Arrow key Up/Down

Folder Browser	Key Command
Edit next tag	Tab
Edit previous tag	Shift+Tab
Edit current tag	Enter
Stop editing tag and discard edits	Escape

8 Online vs. Offline Encoding

We can now discuss the advantages and disadvantages of online encoding with the Plug-In (see section 5) and offline encoding with the Manager application (see section 7).

Offline Encoding is now done with the included desktop application – the Pro-Codec Manager.

Advantages of Offline Encoding:

- Bit stream overloads are consistent,
- The original file length features of the files can be employed (meaning consistent start and end points).
- Batch processing – simultaneously encoding a single file to multiple formats, multiple files to a single format, or multiple files to multiple formats is now possible!

Disadvantages of Offline encoding:

- An extra step in the workflow is required (compared to online encode, where the encoded files are created during the bounce),
- There is no automatic downsampling for offline encode, so the higher sample rate master has to be bounced and downsampled first before working on codec-specific enhancements. With online encoding, the mix is automatically downsampled, and a codec file can be generated directly from this downsampled signal.

Note:
Online encoding does not necessarily add DSP processing as the encoders are already in action. However, additional data buffering and disc writing will add significant overhead, and might cause CPU overloads.

8.1 When to Use Online Encode

In general, offline encoding with the Pro-Codec Manager application should be used when a bounced WAV/AIFF file exists. If there is no WAV/AIFF file (perhaps instead there is a work-in-progress mix) online mode will be quicker and easier, hence more appropriate.

It is possible with some hosts to bounce the mix to wav at the same time as online encoding; the Pro-Codec Plug-In might be inserted into an 'aux' channel for this. When online encoding, the plug-in outputs the unprocessed input signal, so there is no need to bypass it.

If the targets are both a compensated WAV/AIFF and a bit stream for a single codec, then it's best to trim the input level (not the bit stream trim level) and use online encode while bouncing down to WAV/AIFF. An alternative here would be to additionally encode to HD-AAC.

If many different codec files are required with no optimisation for any given codec/bit rate/quality combination, then batch encoding of a bounced WAV/AIFF with the Manager application is appropriate. To ensure the bit streams are perfectly encoded to avoid decoded bitstream overloads, use the **Clip Safe** option and perhaps also the **Normalise** option in the Manager application.

Online encoding should be avoided if many processor-intensive codecs are selected.

9 Specifications

9.1 Plug-In

9.1.1 Delays and CPU Usage

The sample delay (latency) of the plug-in depends upon the particular codecs selected. The monitor outputs are delay compensated, so if more than one codec is selected in the **Codec List**, then the reported delay is the maximum of the individual codec delays. This field is updated in the GUI immediately a new codec or other setting is selected. Depending on the host behaviour, the new value might not be available for delay compensation until the plug-in is removed and re-instantiated.

Individual codec sample delays

Codec	Mode	44.1 kHz	48 kHz	88.2 kHz	96 kHz
MP3	CBR max/min	13849/7345	13849/7345	26978/14690	26978/14690
	VBR	37041	39127	74082	78254
AAC-LC	CBR/VBR	3648	3648	6144	6144
HE-AAC	CBR max/min	7398/7300	7398/7300	7296/3648	7296/3648
	VBR	7300	7300	14600	14604
HE-AAC v2	CBR/VBR	9153	9153	18306	18306
HD-AAC	n/a	4096	4096	6144	6144
Apple AAC	VBR	5120	8192	12288	12288

CPU Use

All codecs are mathematically intense (some more than others), and some have significant latency as a result. The user can take advantage of advances in CPU power in modern computers if appropriate, but can also use the plug-in extensively with older and possibly lower-powered computers by selecting fewer codecs simultaneously.

In particular, the HD-AAC lossless codec creates a heavy CPU load. High sample rates exacerbate this situation; for example, without a powerful CPU, the HD-AAC codec might cause a prohibitively high CPU load at 192 kHz.

All codecs that are added to the **Codec List** are always active, in order to report accurately the NMR signal and potential overload state. This is irrespective of which codec is selected for monitoring. Remember that this requires both encoding and decoding of the signal in real time. Thus it is possible to have dropouts with an underpowered CPU if several highly intensive codecs are active simultaneously.

Online recording will add to the CPU activity (and file-handling activity).

Offline encoding and decoding are CPU-bound, not real time bound.

The graphical area is not CPU-intensive, and should not cause performance problems even on older CPUs.

9.1.2 Internal Buffer Sizes

Additionally, for most codecs, the plug-in uses internal buffer sizes of 1024 for up to 48 kHz operation; 2048 for greater than 48 kHz up to 96 kHz operation; and 4096 for sample rates higher than 96 kHz. We recommend that the buffer sizes of the sound card, and thus of the host application, are set to match these figures.

Buffer sizes for the HE-AAC and HE-AACv2 codecs are twice these figures (so 2048 for up to 48 kHz operation; 4096 for greater than 48 kHz up to 96 kHz operation; and 8192 for sample rates higher than 96 kHz).

If the buffer sizes are not matched, there will be an inefficiency in the data transfer, and in some applications this leads to uneven CPU behaviour.

9.1.3 GUI Controls

Input Meter

0 dB – -63 dB

Graphical Display Frequency

20 Hz – 20 kHz (44.1 and 48 kHz)

Graphical Display Gain

+10 dB – -120 dB (to -180 dB with zoom)

Bitstream Level Trim

-12 dB – 0 dB

Trim Meter

0 dB – +10 dB

Trim Display

-144 dB to +10 dB

9.2 Manager

9.2.1 Metadata Text Encoding

Text encoding formats are a way of describing the internal byte representation of individual characters. The ID3 and iTunes metadata formats strictly define which text encodings are supported, so that any text tags (e.g. Title, Artist) can be read and written by different applications. This section describes how the Manager handles metadata text encoding for the different metadata formats supported by the Manager.

There are 3 different types of text encoding defined in the ID3 and iTunes metadata standards:

Latin–1

This is an extension of ASCII. It is very space efficient, storing only 1 byte per character; however it has very limited character support.

UTF–16

This standard was introduced in 1990 to address universal character support. It is far more comprehensive than latin–1, but is also significantly less space efficient.

UTF–8

This is a later revision of UTF–16. It supports the same number of characters as UTF–16, but has the benefit of being far more space efficient. UTF–8 is generally accepted as the de facto text encoding standard, and is what the Manager uses to store metadata tags internally.

The Manager always writes iTunes metadata text tags using UTF–8 encoding. The handling of ID3 text encoding, however, is more subtle, as it supports Latin–1, UTF–16 and UTF–8 text encoding formats. It is important to note that UTF–8 support is only available in ID3v2.4.

To ensure maximum space efficiency, all text tags less than or equal to ID3v2.3 are written with Latin–1 encoding. However if some characters in the tag are not in the Latin–1 subset, the tag is written as UTF–16. Files with ID3v2.4 text tags will always have the text encoding fixed to UTF–8.

9.3 Codec Specifications

9.3.1 Bit Rate vs Sample Rate

Available bit rates are dependent upon the selected sample rate.

Encoder	MP3
Channel Configurations	Mono, Stereo
Sample Rates (kHz)	8, 11.025, 12, 16, 22.05, 24, 32, 44.1, 48
Bit Rates (kbps)	**Mono**: 8-320 CBR and VBR
	Stereo: 16-320 CBR and VBR

Encoder	AAC-LC, HE-AAC
Channel Configurations	Mono, Stereo, 5.1
Sample Rates (kHz)	**AAC-LC:** 11.025-96
	HE-AAC: 22.05-96
	HE-AAC v2: 22.05-48
Bit Rates (kbps)	**AAC-LC:** 96-256 CBR and VBR
	HE-AAC: 16-128 CBR and VBR
	HE-AAC v2: 18-56 CBR and VBR

Encoder	AAC-LC MPS (MPEG Surround)
Channel Configurations	5.1
Sample Rates (kHz)	**AAC-LC:** 44.1, 48
	HE-AAC: 44.1, 48
Bit Rates (kbps)	**AAC-LC:** 192-320
	HE-AAC: 48-128

Encoder	HD-AAC
Channel Configurations	Mono, Stereo, 5.1
Sample Rates (kHz)	44.1, 48, 88.2, 96, 176.4, 192
Bit Rates (kbps)	Variable (lossless)
Limitations	**Supported oversampling modes:** 1 for input sample rate 44.1 or 48kHz 2 for input sample rate 88.2 or 96kHz 4 for input sample rate 176.4 or 192kHz

9.3.2 Codec and Session Sample Rate

The Plug-In and many of the codecs are sensitive to the session sample rate and channel configuration.

The Plug-In supports a wide variety of host DAW applications, and understandably many of these applications provide differing amounts of session configuration settings to a plug-in. Some applications only present session configuration information to a plug-in when there is an active audio data stream (i.e. the transport is set to play).

As a result of this, care must be taken when:

- selecting codec settings after switching a session sample rate or channel configuration
- when loading a Preset that was saved from a session that used a different configuration

Session loads/restores in Pro Tools, Logic, Cubase and Nuendo are immune from this problem.

It is possible in some applications that a newly instantiated Pro-Codec has not been provided with current session data. If this is the case, when a codec configuration is selected (manually or via a preset load) that is incompatible with the real session configuration, then the Plug-In could receive incompatible data when playback starts. In all cases, the Plug-In will endeavour to match the requested codec settings, and where possible will select the closest compatible settings. By preference, the substitute codec will be of the same type, but using the closest compatible bit rate. Where the codec

doesn't support the sample rate at any bit rate, the codec will default to AAC-LC at the closest bit rate.

To illustrate this behaviour, consider a stereo session at 48 kHz with three codecs selected. If the session configuration is changed to mono at 32 kHz, the plug-in will make the following codec selections:

	Selected Codecs Session Stereo 48 kHz	Substituted Codecs Session Mono 32 kHz	Reason
Codec 1	HE-AAC 128 kbps	HE-AAC 56 kbps	HE-AAC 128 kbps not supported at 32 kHz.
Codec 2	HE-AAC v2 56 kbps	AAC-LC 56 kbps	HE-AAC v2 is stereo only.
Codec 3	HD-AAC 160 kbps	AAC-LC 96kbps	HD-AAC not supported at 32 kHz.

To avoid these problems after changing the session configuration, it is advisable when using affected host applications to press **PLAY**, then **STOP**, before instantiating the plug-in or loading a plug-in Preset. This will ensure that session sample rate and channel configuration data are flushed through the plug-in correctly.

It is also advisable to save Presets for the plug-in at a variety of session sample rates and channel configurations.

To illustrate this further, consider the following examples. WaveLab 7 is used in these examples because it has the powerful ability to easily select and play files with different sample rates. Several other applications exhibit the same behaviour.

Example 1: Different session sample rates and compatible codec bit rates

WaveLab 7 is an example of an application that only provides session configuration information to a plug-in when there is an active audio data stream (i.e. the transport is set to play).

Using WaveLab 7, make three mono 24 bit wav files available via the tabbed windows in the audio file workspace. One is 32 kHz, and is currently selected and playing. The unselected tabs have a 44.1 kHz wav file and a 22.05 kHz wav file available.

A Sonnox Fraunhofer Pro-Codec is instantiated and shows the default codec as AAC-LC 192 kpbs. Note the usual default of 256 kbps is not valid for a mono WAV file (see section 9.3.4 for the full table of codec supported sample rates).

If the 22.05 kHz file is selected, and play stopped and re-started, the codec will automatically select 28 kbps (which is the closest to 192 kbps that the codec supports at 22.05 kHz).

If the 44.1 kHz file is now selected, and play stopped and re-started, the codec will automatically select 56 kbps (the closest to 28 kbps that the codec supports at 44.1 kHz).

Example 2: Different session sample rates and incompatible codec bit rates

Using WaveLab 7, set up two wav files in the tabbed windows in the audio file workspace. One is stereo 24 bit 32 kHz, and is currently selected and playing. The unselected tab has a stereo 24 bit 48 kHz WAV file associated with it.

A Sonnox Fraunhofer Pro-Codec is instantiated. As the HD-AAC codec does not support 32 kHz, the HD-AAC option does not appear in the codec selection window.

The transport is stopped, and the 48 kHz file is selected. The user now tries to select the HD-AAC codec, but it is still not available as an option. After the transport is set to play, the codec can be changed to HD-AAC. The file plays normally and the plug-in and codec perform normally.

The transport is stopped, and the 32 kHz file is re-selected. Now, if the transport is set to play, the codec will receive invalid audio data, and the plug-in will substitute an AAC-LC codec.

Quit and re-start WaveLab 7. Select a 32 kHz wav but do not press play. Instantiate the Pro-Codec. At this moment the plug-in has not picked up the correct session data because WaveLab has not been set to play. The plug-in has been defaulted to 44.1 kHz. Note it is possible now to select HD-AAC but, if this is done, the plug-in will substitute an AAC-LC when playback starts.

Now try to load a preset saved from a 32 kHz session. The preset will be greyed out in the load preset menu, and cannot be loaded. To progress, press **PLAY**, then **STOP**, and then load the Preset.

Summary

These complications arise mostly when low sample rates (32 kHz) are combined with low bit rate codecs; or when high sample rates are combined with high bit rates.

Generally Pro Tools, Logic, Cubase, and Nuendo are among the hosts that are immune to this behaviour, but Sequoia, Samplitude and WaveLab are examples of hosts that can exhibit these issues.

To avoid these problems after changing the session configuration, it is advisable when using affected host applications to press **PLAY**, then **STOP**, before instantiating the plug-in or loading a plug-in preset.

This will ensure that session sample rate and channel configuration data are flushed through the plug-in correctly.

It is also advisable to save presets for the plug-in at a variety of session sample rates and channel configurations. It is advantageous to categorise the presets by channel configuration and session sample rate.

9.3.3 Codec and Encoded File Sample Rates

The sample rate of an encoded file is derived from the session sample rate, the codec supported sample rates, the selected codec bit rate, the selected codec quality setting and the channel configuration. Hence, it is not a simple process to choose a desired encoded file sample rate. The codec sample rate is shown in the codec settings selector.

The following tables give examples of the encoded file sample rate that result from various selection conditions (for full tables, see section **9.3.4** (Preferred Sample Rates)). The codecs have an internal **preferred sample rate**, which will give the highest quality reproduction (but, quite frequently, higher CPU usage).

In general, if the highest quality setting is selected for a codec, then the encoded file will have the sample rate as listed in the **Preferred Sample Rate** column (matching the session sample rate, if this option is available).

In all other cases of quality setting, a codec uses the same sample rate as the session, except at higher sample rate sessions when the codec will use the highest simple division of the session sample rate that the codec supports.

MP3 stereo, session 44.1 kHz or 88.2 kHz

Bit Rate (kbps)	Supported Sample Rates (kHz)	Preferred Sample Rate (kHz)	Quality Setting	Actual Sample Rate Used (kHz)
96	32/44.1	32	Fast	44.1
96	32/44.1	32	Highest	32
128	32/44.1/48	44.1	Fast	44.1
128	32/44.1/48	44.1	Highest	44.1
320	32/44.1/48	44.1/48	Fast	44.1
320	32/44.1/48	44.1/48	Highest	44.1

MP3 stereo, session 48 kHz or 96 kHz (96 kbps not supported at this sample rate)

Bit Rate (kbps)	Supported Sample Rates (kHz)	Preferred Sample Rate (kHz)	Quality Setting	Actual Sample Rate Used (kHz)
96	32/44.1	32	Fast	n/a
96	32/44.1	32	Highest	n/a
128	32/44.1/48	44.1	Fast	48
128	32/44.1/48	44.1	Highest	44.1
320	32/44.1/48	44.1/48	Fast	48
320	32/44.1/48	44.1/48	Highest	48

9.3.4 Preferred Sample Rates

The supported session sample rates of the plug-in are from 32 kHz minimum to 192 kHz maximum (where the host DAW allows). The supported codec sample rates are from 8 kHz minimnum to a maximum of 48 kHz (MP3), 96 kHz (AAC family) or 192 kHz (HD-AAC only, see above).

Under certain settings at higher session sample rates, downsampling occurs between the session and the audition path, or between the session and the encoder. This downsampling is lossless to 24 bits.

The following tables indicate the preferred sample rates of the codecs (see section 9.3.2). Only supported sample rates from 32 kHz and higher are shown.

MP3

Bit Rate Range (kbps)	Supported Sample Rates	Preferred Sample Rate	Channel Count
40 – 47	32	22.050	1
48 – 55	32	24	1
56 – 73	32/44.1/48	32	1
64 – 320	32/44.1/48	44.1	1
96 – 111	32/44.1	32	2
112 – 319	32/44.1/48	44.1	2
320	32/44.1/48	44.1/48	2

AAC-LC

Bit Rate Range (kbps)	Supported Sample Rates	Preferred Sample Rate	Channel Count
32 – 55	32	32	1
56 – 127	32/44.1/48	44.1	1
128 – 191	32/48/96	48	1
192 – 264	44.1/48/96	96	1
264 – 288	48/88.2/96	96	1
40 – 95	32	32	2
96 – 111	32/44.1/48	32	2
112 – 255	32/44.1/48	44.1	2
256 – 384	32/48/96	48	2
384 – 529	44.1/48/96	96	2
529 – 576	48/88.2/96	96	2
160 – 239	32	32	5, 5.1
240 – 279	32/44.1/48	32	5, 5.1
280 – 639	32/44.1/48	44.1	5, 5.1
640 – 800	32/48/96	48	5, 5.1

HE-AAC

Bit Rate Range (kbps)	Supported Sample Rates	Preferred Sample Rate	Channel Count
12 – 17	32	32	1
18 – 39	32/44.1/48	44.1	1
40 – 56	32/44.1/48	48	1
16 – 27	32/44.1/48	32	2
28 – 95	32/44.1/48	44.1	2
96 – 128	32/48/96	48	2
64 – 69	32/44.1/48	32	5, 5.1
70 – 159	32/44.1/48	44.1	5, 5.1
160 – 319	32/44.1/48	48	5, 5.1

HE-AAC v2

Bit Rate Range (kbps)	Supported Sample Rates	Preferred Sample Rate	Channel Count
12 – 17	32	32	2
18 – 39	32/44.1/48	44.1	2
40 – 56	32/44.1/48	48	2

AAC-LC MPS (MPEG Surround)

Bit Rate Range (kbps)	Supported Sample Rates	Preferred Sample Rate	Channel Count
192	44.1/48	44.1	5.1
256	44.1/48	44.1	5.1
320	44.1/48	44.1	5.1

HE-AAC MPS (MPEG Surround)

Bit Rate Range (kbps)	Supported Sample Rates	Preferred Sample Rate	Channel Count
48	44.1/48	44.1	5.1
64	44.1/48	44.1	5.1
96	44.1/48	44.1	5.1
128	44.1/48	44.1	5.1

10 Commercial use of MP3 Content

It is important to distinguish between the use of codecs and the use of content produced by the codecs. Therefore it is important to note that any MP3 royalty fees are independent of which encoder is used to produce the MP3 file content. If royalty fees are due for a particular project or enterprise, they will be due irrespective of whether the Pro-Codec is used or some alternative encoder is used. **There is no circumstance where using the Sonnox Fraunhofer Pro-Codec will increase licence fees or require additional fees of any sort.**

Both the MP3 and AAC compression routines are based on technology that is patented. At present no royalties are required for AAC codec usage, and private or non-commercial use of MP3 content does not require royalty fees. Under some circumstances the commercial (i.e. revenue-generating) use of MP3 content will require a separate licence irrespective of how that content is generated. To summarise:

- There are no licence fees for private or non-commercial use of MP3 content.
- Enterprises with a turnover of less than $100,000 per year are exempt from MP3 royalty payments.
- There are currently no royalties collected for AAC commercial content distribution.

Note:
If you are in doubt about whether fees are applicable for your project, please contact www.mp3licensing.com

11 Supported Platforms

- Avid Pro Tools (AAX Native 32/64-bit)
- VST hosts (32/64-bit)
- AU hosts (32/64-bit)
- Mac Intel OSX 10.6 or higher
- Windows 7 and 8 (32/64-bit)

12 System Requirements

For latest System requirements, please visit www.sonnox.com.

All versions

- Free iLok account
- Appropriate product licence
- iLok2

Pro Tools

- Approved Digidesign/Avid CPU and hardware configuration
- Pro Tools 10.3.5 (Native or HD), or higher

VST Native

- VST compliant host application (e.g. Cubase, Nuendo, etc.)

Audio Units

- Approved Apple CPU and OSX 10.6 or higher
- Audio Unit Host application (e.g. Logic, Digital Performer, etc.)

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